PortaNet

User Manual
Ver. 5.4.1.2

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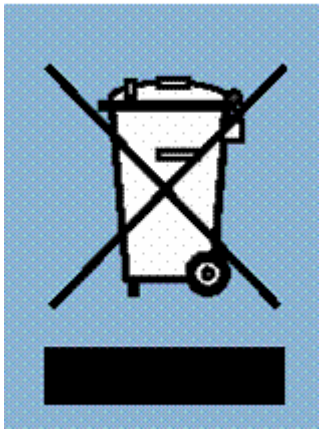
CE Declaration of Compliance

Procesamiento Digital y Sistemas S.L., hereby declares that PortaNet bearing the CE168X marking is in compliance with Electromagnetic Compatibility Directive (89/336/EEC), and the Low Voltage Directive (72/23/EEC) of the European Union.

A "Declaration of conformity" for PortaNet is available on file at Prodys offices in Spain. To obtain this information, please email sales@prodys.net.

CAUTION

PortaNet uses a Lithium battery.
Danger of explosion if battery is incorrectly replaced. Replace only with the same or equivalent type recommended by the manufacturer. Dispose of used batteries according to the manufacturers instructions.



Your product is designed and manufactured with high quality materials and components, which can be recycled and reused.

When this crossed-out wheeled bin symbol with black bar underneath is attached to a product it means that product is covered by the European Directive 2002/96/EC.

Please, inform yourself about the local separate collection system for electrical and electronic products.

Please act according to your local rules and do not dispose of your old products with your normal household waste. The correct disposal of your old product will help prevent potential negative consequences for the environment and human health.

About this manual

This manual is the installation and operation guide for Portanet. With this manual, the user will gain a fundamental understanding about the main features and operation modes.

This manual is arranged in the following manner:

1. PortaNet general specifications.
2. Installing the PortaNet.
3. Operation guide.
4. Remote control.
5. PortaNet blocks.
6. How it works.
7. Examples of applications.
8. Problem-solving guide.
9. Technical specifications.
10. Connectors.
11. Updating the firmware.

General specifications

PortaNet is the first really portable audio codec which provides real time audio communications over IP networks. In addition, it comes fitted with an 1-BRI ISDN interface, which allows ISDN connectivity at those places where IP is not available.
PortaNet, as a member of the Prodys IP audio codec family, includes all the features of the ProntoNet, but adds some important enhancements in the audio interfaces and operationally for portable use.

These are the PortaNet most important features:

- **Analog Audio:** PortaNet incorporates a mixer with four MIC/LINE level configurable audio inputs and an extra line level input that can be mixed to the headphone outputs to provide the international sound to the commentators. It also includes three headphone outputs and two line level analog balanced audio outputs. All the outputs can be configured by the user to suit the audio signal to be carried, to adapt to the needs of different applications.

- **Compression algorithms:** PortaNet is equipped with the widest range of compression algorithms without any additional cost: G711, G722, PCM, MPEG1,2 Layer II, Mpeg 1,2 Layer III, MPEG 2,4 AAC LC, MPEG4 AAC LD, MPEG4 AAC HE, Standard and Enhanced apt-X™.

- **Communications**: IP and ISDN connectivity. With PortaNet it is possible to establish two independent connections, one for program and the other for coordination.

- **Power supply**: Desk Top AC/DC Power Converter and an optional battery which provides 3 hours of autonomy. The battery can be charged on the system and its status can be monitored on the screen.

- **Auxiliary data**: PortaNet includes a serial port (RS232) and a GPIO port with 2 inputs and 2 outputs. The auxiliary data can be sent/received along with the audio for remote control/signalling.

- **Control and monitoring**: The unit can be configured from its front panel and from its embedded web page. All the controls are configurable remotely. The keypad can be controlled and also blocked from the web page for security reasons.

- **Small and lightweight**: its weight is about 2 kg. and its dimensions are: 70 x 285 x 243 mm.

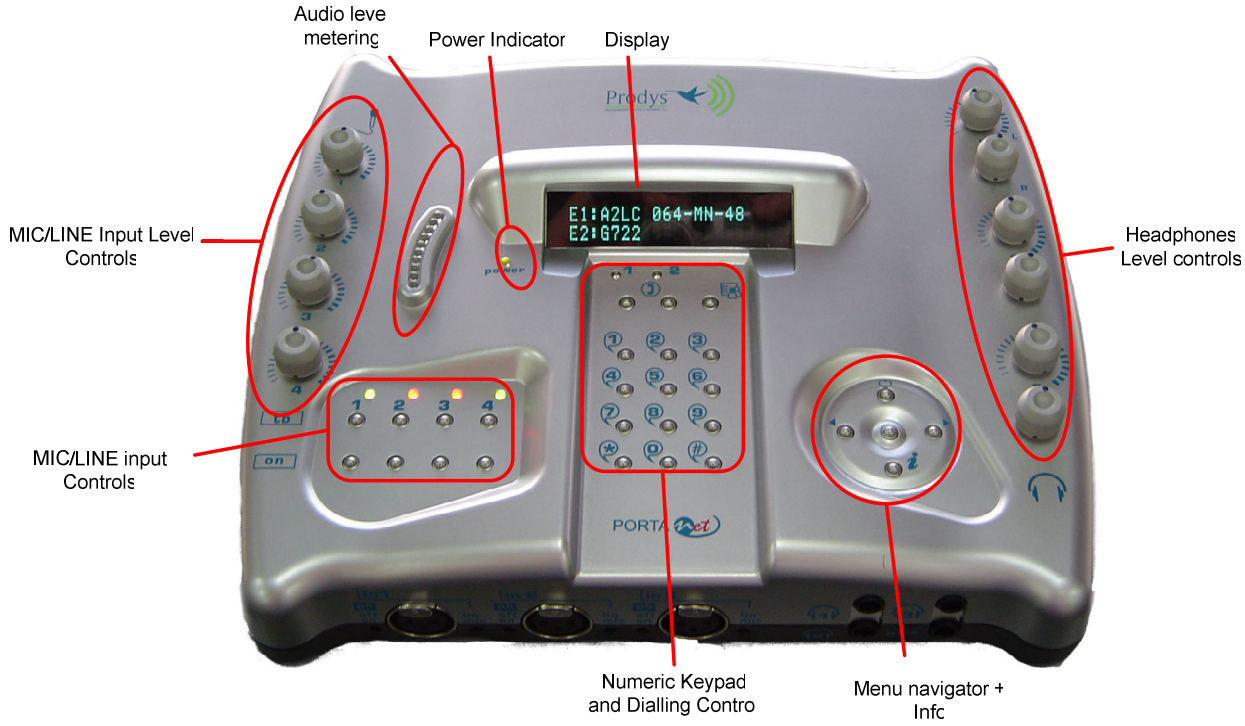
Installing the PortaNet

Before unpacking unit check its packaging for any signs of damage or mishandling during transportation, report any damage to the shipping company immediately. Unpack the unit carefully, if you find any damage or the unit does not work correctly, you should contact Prodys or its distributor as soon as possible.

III.1 PortaNet overview

PortaNet has all its connections in its front and rear panels. The keypads and display are placed in the top surface.

III.1.1 Control Panel



Audio Level controls for the MIC/LINE inputs

Each audio input can be adjusted independently. The audio level control can be made locally or remotely by using the PortaNet web browser.

Audio Level controls for the headphone outputs

The headphone audio level can be adjusted independently for each ear. It is possible to monitor different signals on either side of the headphones.

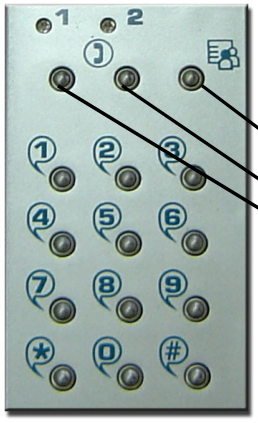
Menu Keypad

Navigation Keys: The keys ⇐ , ⇑ , ⇒ , are used for moving around the menus and the **OK** is for selecting/accepting the desired action or parameter.

Inf key

This key allows the user a simple and quick way to display detailed information on the status and configuration of the PortaNet.

Call management keypad.



The numeric keypad is for entering information such as the IP address or ISDN number that you wish to connect to.

- BOOK for establishing communications using a configuration saved in the Address Book.
- CALL1 and CALL2 are for establishing and terminating connections, and also to monitor the called/calling number.

Input audio control keypad.

- On Key → Each audio input has a key to enable/disable it. When an audio input is enabled, it is connected to the program line (line 1).

- Tb Key → PortaNet allows the user to establish two independent communications, using either IP or ISDN. Each input can be configured to be sent through the program line (line 1) or through the coordination or TalkBack line (line 2). Pressing the Tb key for one input causes this input to be mixed with those signals assigned to the coordination or TalkBack line (line 2).

LED´s → There is a led related to the status of each input:

- OFF: Input disabled.
- Green: Input enabled and connected to the program line (line 1).
- Red: Input enabled and connected to the TalkBack line (line 2).

Power LED

It shows the power supply status and also battery status.

-GREEN: The external power converter is powering the unit. The battery is not connected.
-BLINKING ORANGE: The external power converter is powering the unit. The battery is connected and it is being charged on the system.
-ORANGE: The external power converter is not powering the unit. The battery is connected and powering the unit.
-BLINKING RED: The external power converter is not powering the unit. The battery is connected and powering the unit, but it is about to run out (about 20 minutes / 10% of battery capacity).

When the unit is starting, the Power LED will be blinking green.

III.1.2 The front panel and its connectors

In the front panel there are two areas: Audio inputs and Headphone outputs. Because of space restrictions the Auxiliary output level control is also there.

> ***There is an additional fourth input in the rear panel. This input doesn't have an associated headphone output.***

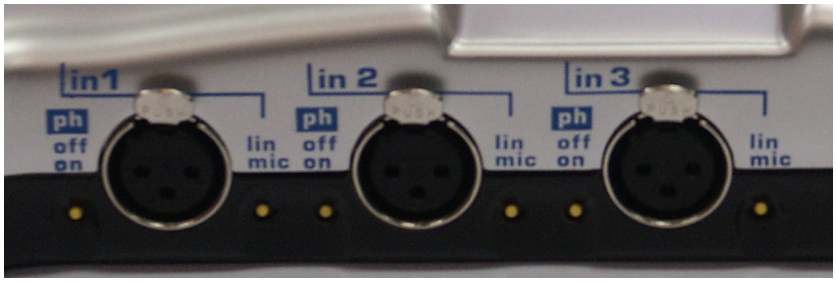
MIC/LINE Audio inputs: There are four MIC/LINE audio inputs. Each audio input includes a switch to select MIC or LINE level, and another one for selecting whether to supply 48 volts. phantom power to the MIC.

line/mic switch:

- When this switch is in the UP position, the line level input is selected (maximum input level +20 dBu).
- When this switch is in the DOWN position, this input is configured to work as a MIC level input.

ph swtich(48 volts. Phantom powered):

- When this switch is in the UP position (OFF), this option is disabled.
- When this switch is in the down position (ON), this option is enabled.



Headphone outputs: There are three headphone outputs, each one related to its corresponding audio input. Signals present on the headphone audio outputs will be dependent on the configuration of the audio headphone matrix available in the PortaNet web browser. More information about it in next chapter about PortaNet's input and outputs.
There are audio level controls for each output and for each ear.

Line level balanced auxiliary audio output: Along with the three headphone outputs, there is a fourth output called 'aux'. This audio output is totally independent from the headphone outputs, and it is placed together with these only for space saving reasons. This output is a balanced line level output, with a

maximum level of +20 dBu. This output is user configurable; that means that it is possible to define which audio signals will be present in this output.

Further information about audio output configuration can be obtained from chapter 5 - PortaNet Blocks.

III.1.3 Rear Panel

The rest of connections are located on the rear panel:

III.1.3.1 Fourth MIC/LINE audio input (in 4).

The same as the inputs from the front panel, it can be configured as LINE or MIC level, and it has the possibility to be phantom powered (48 volts.) However, it doesn't have a headphone output related. For this purpose, the 'aux' output might be used.

III.1.3.1.1 LINE level input (International Sound Input - Int. in)

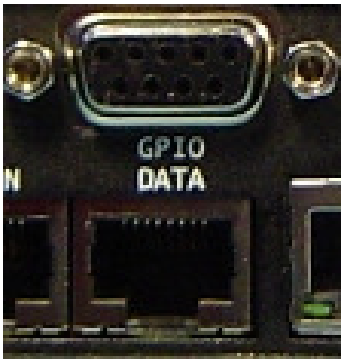
This is an additional line level input which can be mixed with the headphone outputs to provide the international sound to the commentators.

III.1.3.2 Second auxiliary line level output.

Called REC output, this output can carry the program line signal, the return from the program line or the sum of both signals. This is a balanced XLR connector.

Further information about audio output configuration can be obtained from chapter 5 - PortaNet Blocks.

III.1.3.3 Auxiliary data: GPIO and RS232 ports.



The GPIO port on a DB9 connector allows remote control/signalling by means of remote contact closures. There are two ground contact inputs and two relay outputs.

Another DB9 connector at the rear panel provides an RS232 port for sending/receiving auxiliary data along with the encoded audio for IP and ISDN communications.

Over ISDN, the auxiliary data and GPIO information will be only available in those compression algorithms which support ancillary data along with the encoded audio, and when this option is enabled.

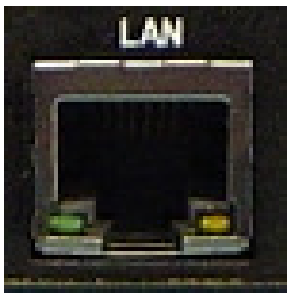
Over ISDN, it is possible to choose between ProntoNet format and Pronto2/3 format (in which the GPIOs information cannot be sent). To be backward compatible with ISDN Pronto2/3 Prodys devices, select Pronto2/3 format.

When selecting IP as communication interface, the auxiliary data can be sent via a different path, different from the audio one. This method has 3 big advantages: less delay independent from the audio codification delay; the possibility to send/receive auxiliary data regardless of the compression mode used for the audio communication; and the possibility to send/receive the User bit of the AES/EBU frame. The drawback is that the audio and data delay won't be the same.

III.1.3.4 Ethernet port – the LAN Connector

The LAN socket is an standard 100Base-Tx (10/100 Mbps) Ethernet connection that takes a RJ45 plug. Through this Ethernet port it is possible to transmit and receive audio, as well as manage the equipment. Next to the socket there are three LEDs that indicate different states for the connection and these are very useful in problem-solving situations.

LAN LED's:

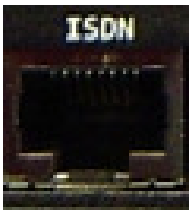


- Green LED → LINK STATUS: ON = Connected
- Orange LED → RECEIVE STATUS: On = Receiving Data.

Further information can be obtained in the chapter - XIV - Connectors.

From the web interface and the front panel menu it is possible to set the speed and duplex configuration to the following values: AUTO, 10HD, 10FD, 100HD, 100FD[1].

III.1.3.5 ISDN Port



The PortaNet incorporates an ISDN terminal adapter that allows connection to a basic rate ISDN line (2B+D). It supports different ISDN protocols (EURO_ISDN, DMS100, AT&T 5ESS and NAT1). There is one RJ45 connector for connecting to an S/T interface S/T (S_0).

There is available a special version with U ISDN interface.

III.1.3.6 Power supply

PortaNet comes supplied with a desk top AC/DC converter (16 volts. output) to be connected to the power connector located at the rear panel. The power converter works with an AC input range of 100 to 240 VAC, 50 to 60 Hz.
The power switch is located just below the power connector.
In addition, there is also available an optional custom battery pack. Whenever the external power converter is connected, the power is supplied from it, and the battery will be charged when necessary. When the external power converter is unplugged from the unit, the battery will take over.

[1] This option is available from version 5.2.1 onwards.

The power LED on the upper panel informs about the status of the battery and the power supply. These are the possible status for this led:

-GREEN: The external power converter is powering the unit. The battery is not connected.

-BLINKING ORANGE: The external power converter is powering the unit. The battery is connected and it is being charged on the system.

-ORANGE: The external power converter is not powering the unit. The battery is connected and powering the unit.

-BLINKING RED: The external power converter is not powering the unit. The battery is connected and powering the unit, but it is about to run out (about 20 minutes / 10% of battery capacity).

Once the unit is switched on, the boot sequence will start. This sequence will take about 50 seconds to finish, when the screen will display the status display. When the unit is starting, the Power LED will be blinking green.

PortaNet audio inputs

IV.1 Location

There are four MIC/LINE level inputs, switchable between LINE and MIC level. Three of them are located on the front panel, and the fourth one is on the rear panel. Moreover, there is an additional line level input located on the rear panel which can be mixed to the headphone outputs.

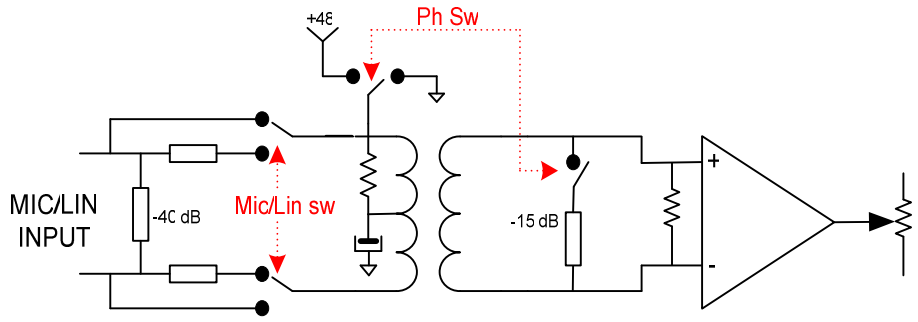
IV.2 Technical specifications

IV.2.1 MIC/LINE level inputs

- All the MIC/LINE level inputs are transformer balanced on a XLR female connector. These inputs are switchable between MIC and LINE levels by means of a switch placed on the right side of the input connector.
- It is possible to supply phantom power (48 volts.) to each input. There is a switch on the left side of the input connector for this purpose.
- When the audio inputs are configured as MIC level inputs, sensibility range is –60 to –25 dBu.

When phantom power is enabled, the whole gain is decreased by 15 dB to compensate for the the higher output of condenser MICs. Thus, the gain range will go from –45 to –10 dBu.

- When the inputs are configured as LINE level inputs, the maximum input level will be +20 dBu.
- Each input level can be adjusted independently.



IV.2.2 International Sound LINE level input.

- Electronically balanced on a XLR female connector.
- Maximum input level is +20 dBu

IV.3 Audio Level Controls

Each audio input can be adjusted independently. The audio level control can be made locally from the knobs or remotely by using the PortaNet web browser.

Once the audio input gain setting has been selected to be controlled by remote control, the PortaNet reminds this effect on its display upon any attempt to increase or decrease the audio gains locally by means of the audio control knobs.

The International Sound Input level can be adjusted only remotely.

IV.4 Audio input configuration

IV.4.1 ON keys

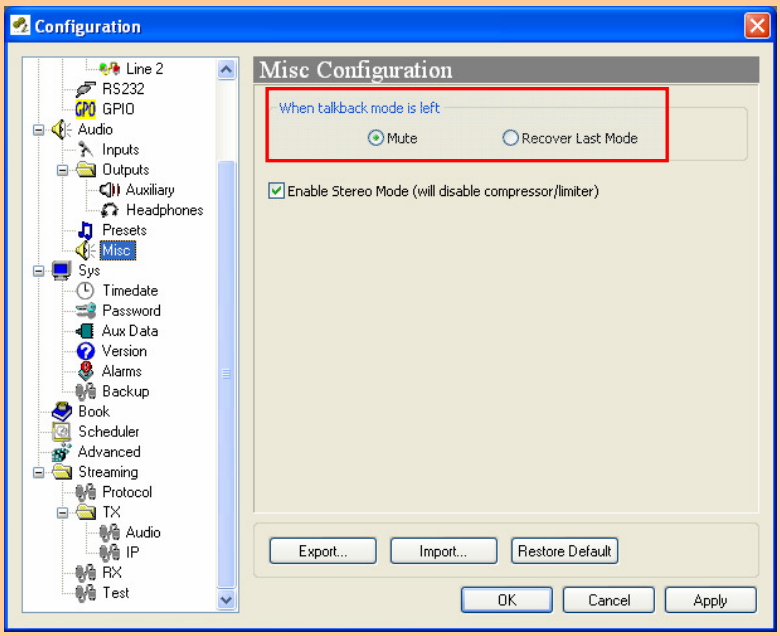
Each audio input has a key to enable/disable it. When an input is enabled, it will contribute to the audio sent through the program line (line 1). The corresponding LED will light on GREEN, indicating that the input is enabled and going through the line 1.

IV.4.2 TB Keys

PortaNet supports two independent communications via IP or ISDN. Thus, we can establish one connection for the main program, and another one for coordination, or TalkBack program. Each input can be configured to be added to the audio contribution on line 1 or line 2 by means of the 'TB' and 'ON' keys located on the top surface.

- When an input is connected to line 2 or TalkBack program (pressing the Tb key) it is disconnected from the program line (line 1). The corresponding LED will light on RED.
- When an input is connected to line 1 or program (pressing the ON key) it is disconnected from the TalkBack line (line 2). The corresponding LED will light on GREEN.
- When the input is disabled, the led is off.

When one input leaves the TalkBack line (line 2), this input is disabled for security reasons. This way, this input cannot enter the program line unintentionally. However, it is possible to change this operation mode from the web page. When the Recover Last Mode is selected, the input will recover its previous status when it leaves the TalkBack line.



PortaNet audio Outputs

There are three headphone outputs and two general purpose balanced audio outputs (line level).

V.1 Location

- The three headphone outputs are located together on the front panel.
- One of the balanced audio outputs is located on the front panel, called 'aux', and the other is on the rear panel, and it is called 'REC'.

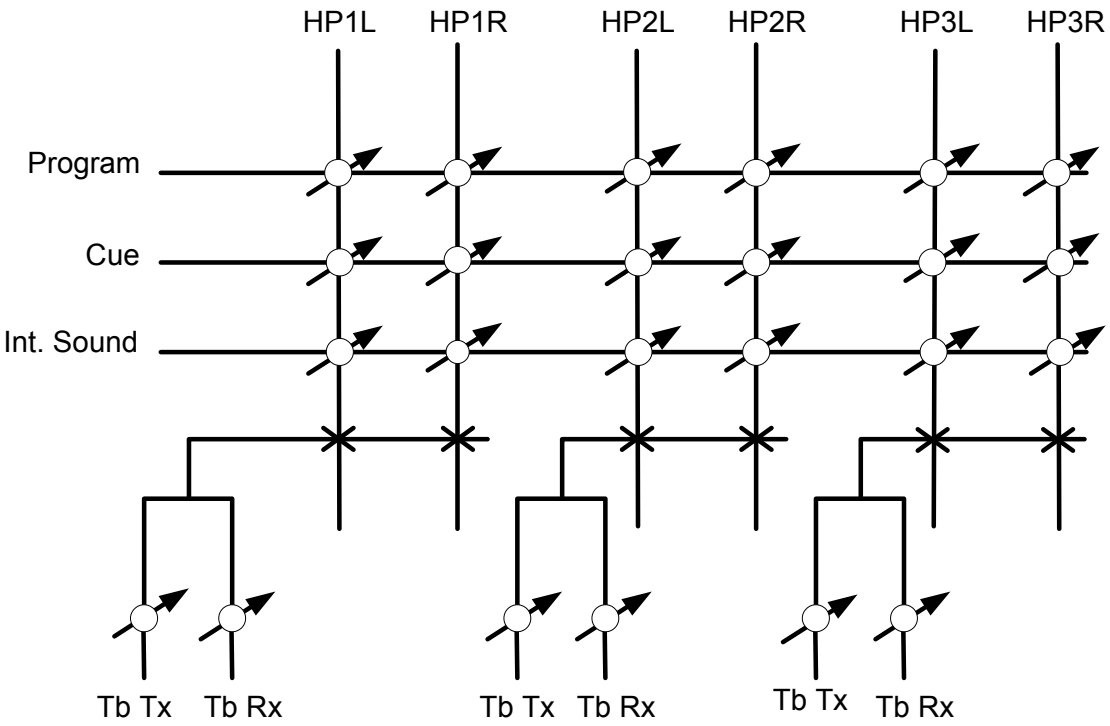
V.2 Headphone outputs.

V.2.1 Technical specifications

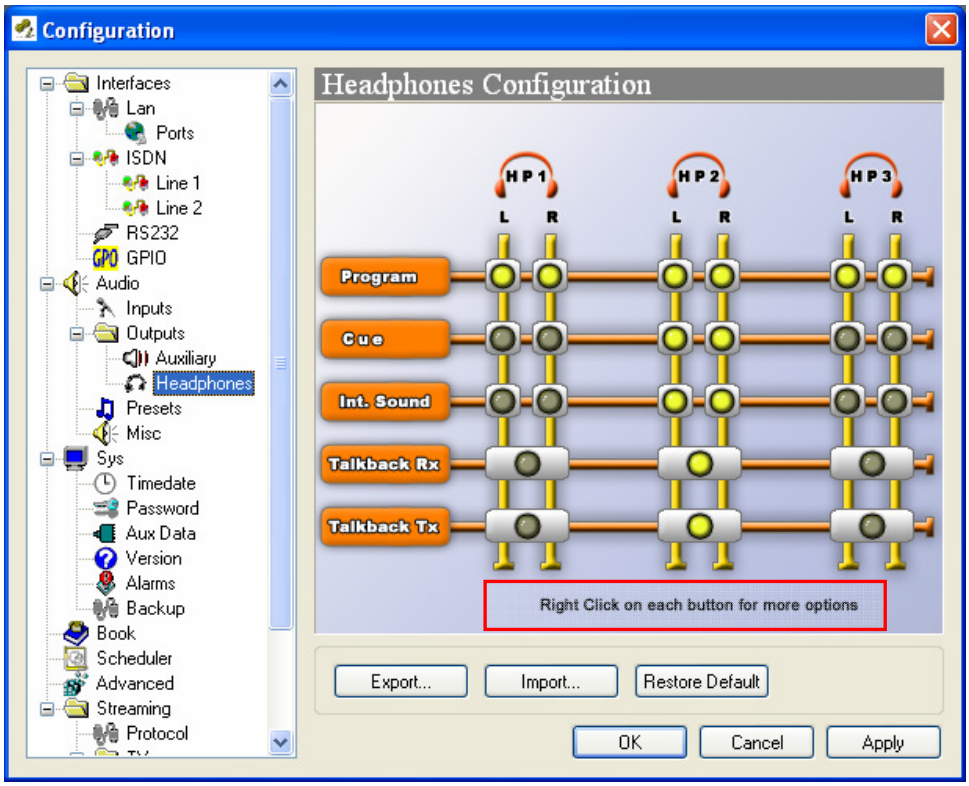
- 6.3mm Jack connector
- Headphone impedance range: 300 – 2K ohms.
- Output Impedance < 25 ohms.
- Bandwidth 20Hz-20kHz (1dB).
- Maximum output level: 6dBu (8 mW) on 300 ohms.
- THD+N < 0.01% @ 1kHz.
- SNR < 95 dB.
- Independent audio sources (local o return) for left and right.
- Independent level adjustment for each output.

V.2.2 Operation mode - Headphones Matrix

- The audio sources for the headphones can be selected from the PortaNet web browser. At the same time, it is possible to define if some of them will be attenuated depending on which line (program or talkback) its corresponding input t is connected.
- The audio sources that can be selected are: Programa, Cue, Talkback Tx, Talkback Rx and International Sound.
- Program, Cue and International Sound can be selected in each ear headphone independently.
- The following diagram shows how can be configured the headphone outputs:

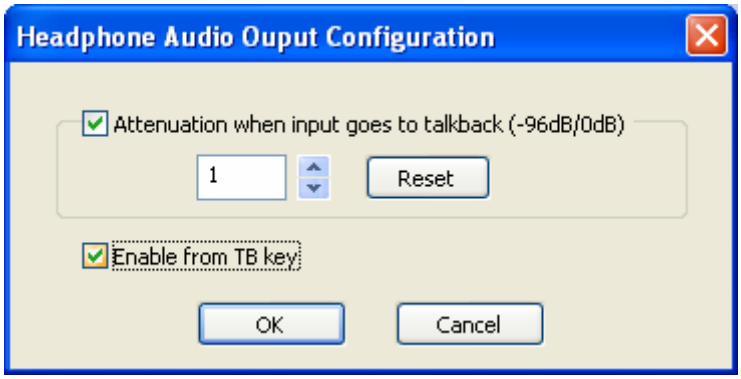


Following is showed the dialog to select the headphones configuration.



By right clicking on each cross point a new dialog appears that allows to define the behaviour of the selected output depending on where the corresponding input is connected, that is, depending on if the MIC/LINE level input is connected

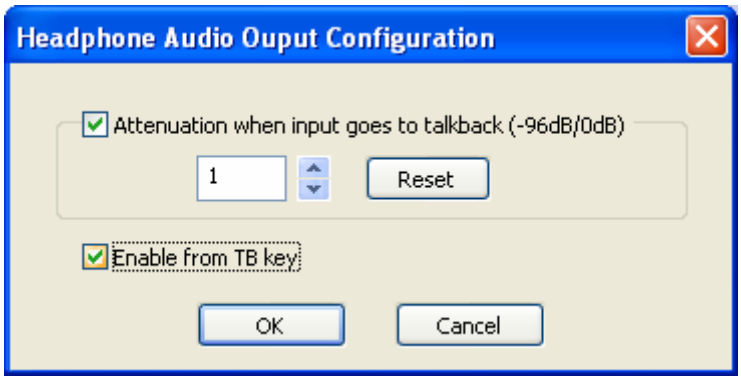
to the program line or to the talkback line. Also, it allows the user to select whether the signal will be controlled from the audio input keypad.



V.2.2.1 Attenuation when the MIC/LINE level input goes to Talkback or Program

It is possible to define an automatic attenuation on signals that feed the headphone outputs when the corresponding inputs go from Program to Talkback or vice versa. The attenuation will be carried out according to the following critera:

Program, Cue or International Sound can be attenuated when the corresponding input is connected to the Talkback channel. It is possible to define an independent attenuation level for either of them.



The same thing can be done regarding Talkback Tx and Talkback Rx, that is, either signal can be attenuated when the corresponding input goes to program.

V.2.2.2 Enable from TB Key

"Enable from Tb Key" is an additional option that only is available for Talback Tx and Talkback Rx. When it is enabled, the corresponding signal will be sent to the headphone outputs only when the corresponding input has been connected to the Talkback channel, that is when its corresponding Tb key has been pressed.

When this option is disabled, this signal will be always available on the headphone outputs regardless of whether the corresponding input is connected to the Talkback channel or not.

V.3 Line level balanced audio output: REC

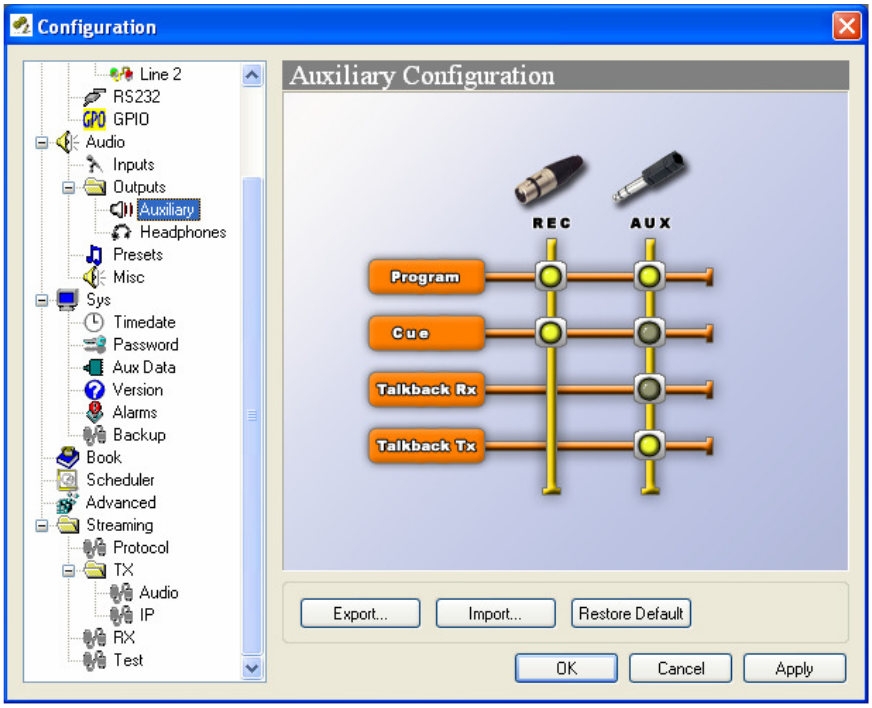
The audio sources for the REC output can be selected from the PortaNet web browser.

V.3.1 Technical specifications

- Bandwidth 20Hz-20kHz (0.3dB).
- Maximum audio level: +20dBu.
- THD+N < 0.005% @ 1kHz.
- SNR < 100 dB.
- XLR male connector fitted to the rear panel.
- In normal operation, this output carries the audio coming from the program line (line 1). This audio signal is called CUE.
- Output impedance < 50 ohms.

V.3.2 Operation Modes

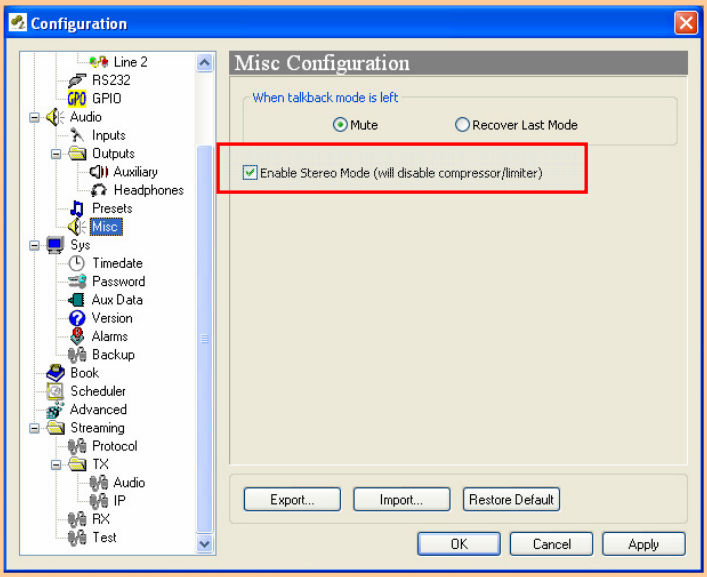
The REC output supports two different configurations depending on the selected audio sources in the auxiliary output configuration matrix of the web browser:



V.3.2.1 Codec Mode

In this mode the REC output carries the audio signal coming from the line 1 or CUE. This mode is selected by clicking Cue signal in the configuration auxiliary output matrix.

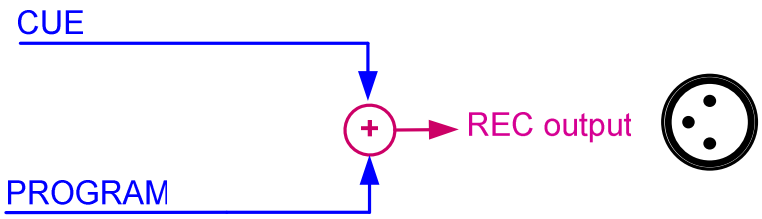
When the REC output is configured in codec mode, it is also possible to work in stereo. In that case, the REC output carries the left channel audio of the received stereo signal. This operation mode requires to be enabled from the web browser in order to disable the compressor/limiter. For more information please read the chapter XI - Operation examples.

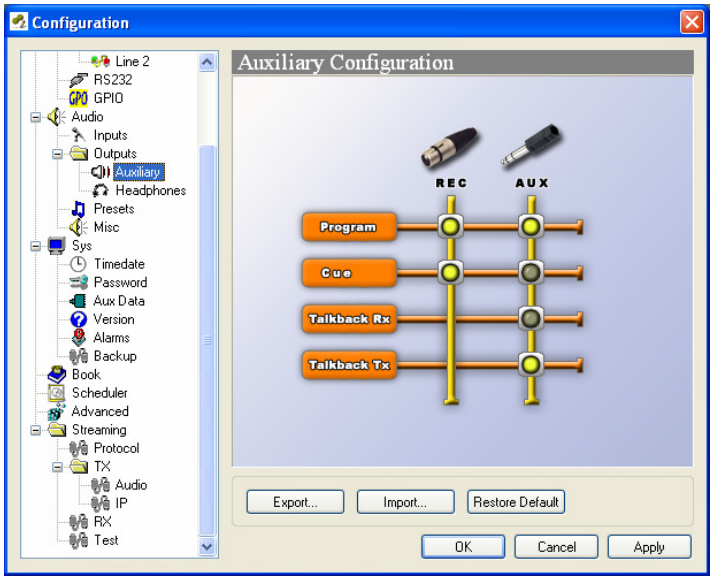


V.3.3 RECORDING operation mode

In this mode, it is possible to add the audio contribution signals on the program line to the return signal from the same line (CUE signal). One application might be to record an interview or a retransmission.

This mode is selected by clicking Cue and Program signals in the auxiliary output configuration matrix.





V.4 'AUX' output.

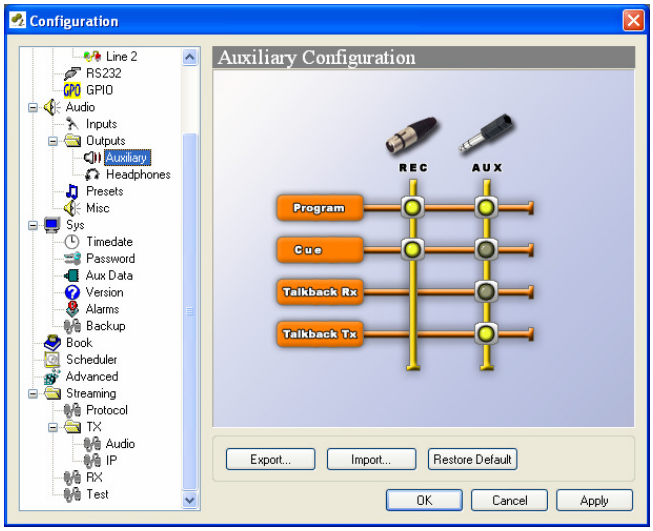
This output supports many different configurations depending on the selected audio sources in the auxiliary output configuration matrix of the web browser:

V.4.1 Technical specifications

- 6mm (1/4") jack connector at the front panel.
- Bandwidth 20Hz-20kHz (0.3dB).
- Maximum output level: +20dBu.
- THD+N < 0.005% @ 1kHz.
- SNR < 100 dB.
- Output impedance < 50 ohms.

V.4.2 Operation modes

Depending on which signals are selected from the auxiliary output configuration, the AUX output supports different configurations.

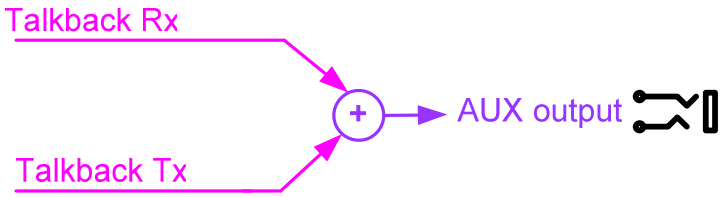


V.4.2.1 CODEC mode

In this operation mode, this output contains the signal coming from the second line (Talkback Rx).

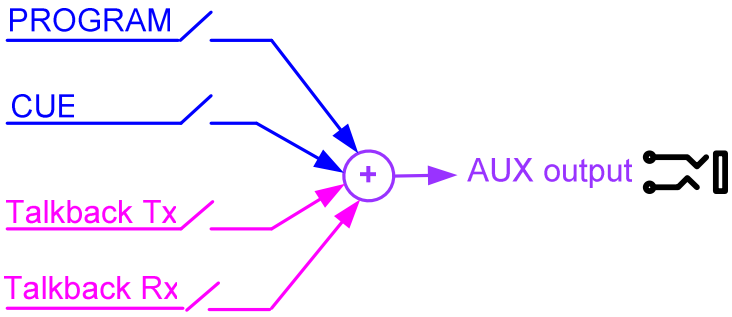
V.4.2.2 INTERCOM mode

In this mode the audio signal going through line 2 (Talkback Tx) is added to the AUX output.



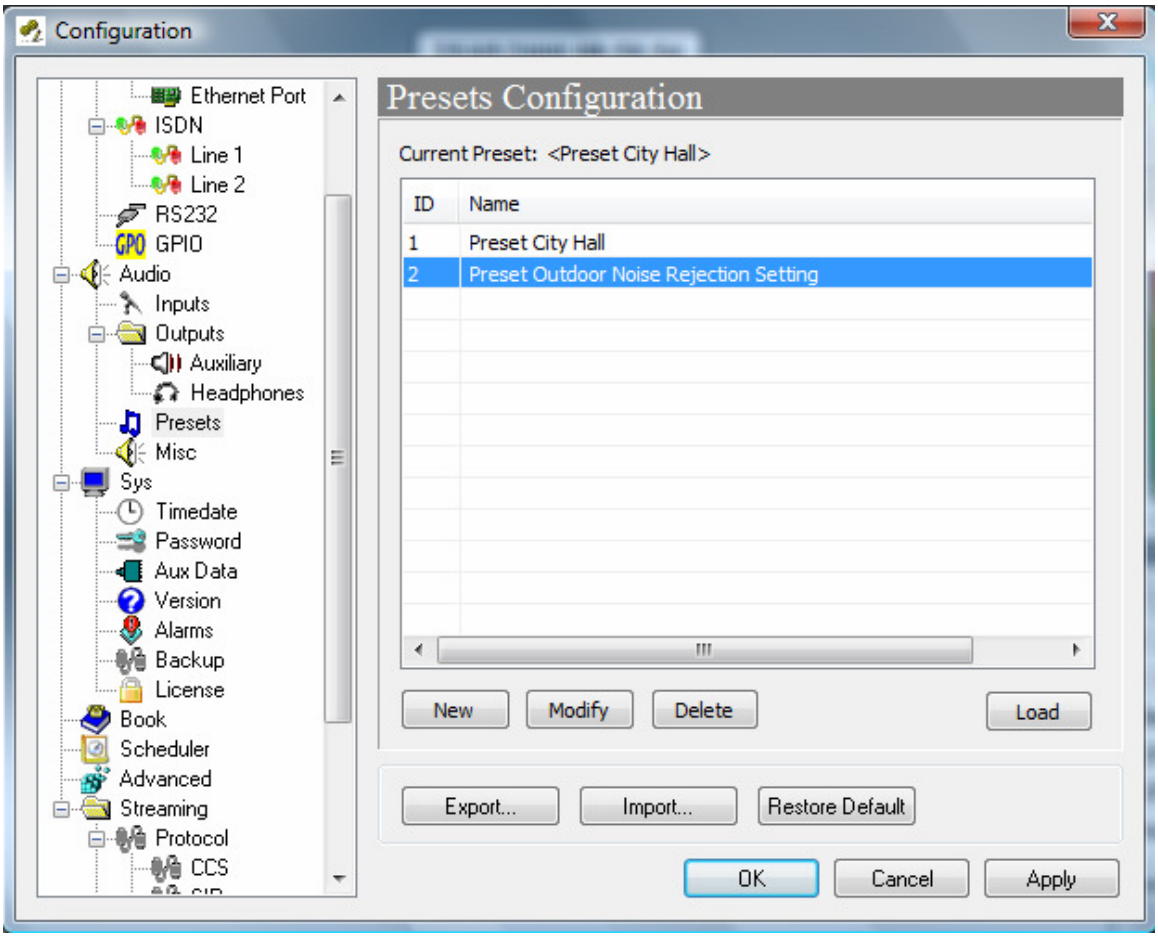
V.4.2.3 Other configurations for the Intercom mode

This diagram shows all the signals which can be routed to the 'aux' output:



PortaNet audio presets

PortaNet Audio presets allow the user to save audio settings and reapply them in a single step.
Audio presets are available in the Audio configuration menu of the PortaNet web browser:



VI.1 What is an audio preset?

An audio preset contains all the configuration of the following items:

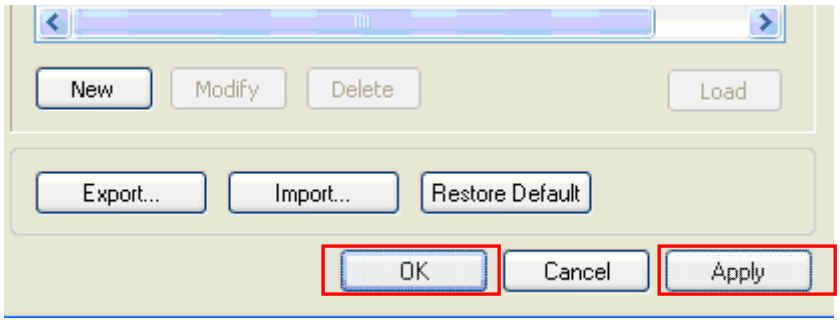
- Audio inputs: It includes whether the audio input levels are remotely controlled or not and, if so, the audio input level configuration.
- Audio Auxiliary output: the configuration for each audio Auxiliary output (REC and AUX outputs).
- Headphone outputs: the configuration of the headphone outputs according to the headphone audio matrix configuration.

- Miscellaneous options: the configuration of the different options available in the Misc Configuration menu.

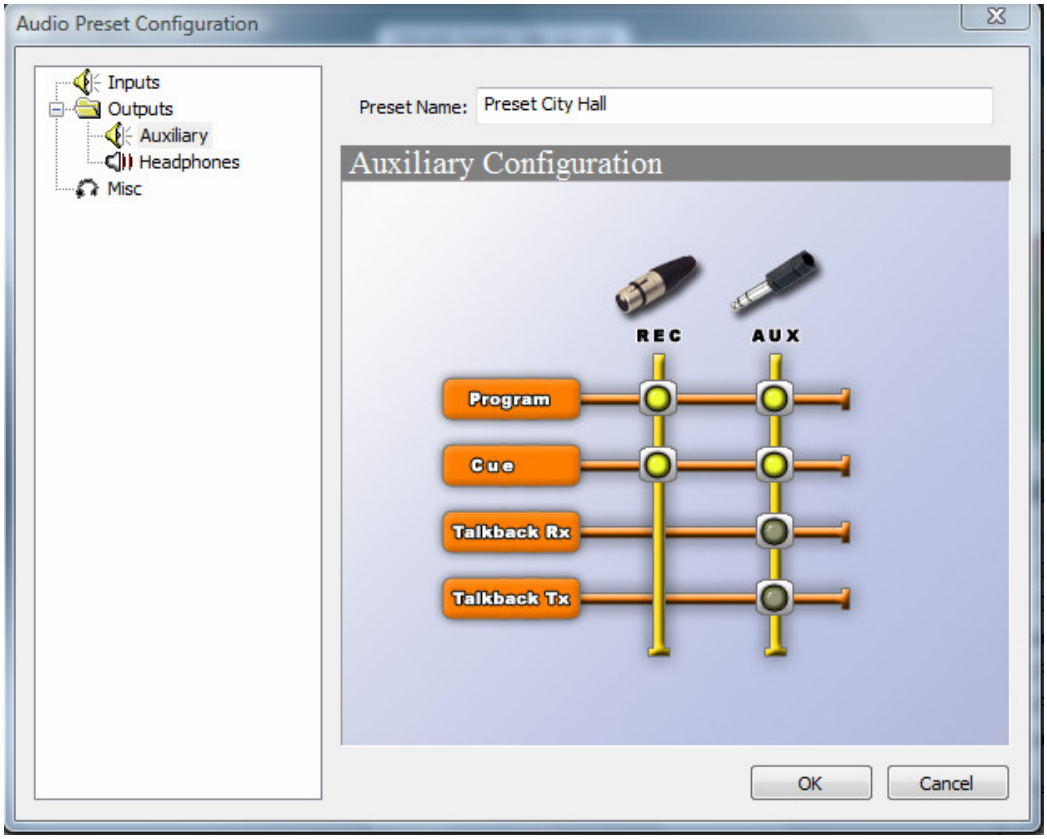
VI.2 Creating & Modifying a preset

To create an audio preset, open the Presets window in the Configuration menu and the click on the "New" button. A new pop up window makes possible to save the current settings and any further modification and to store with the Preset label of your choice (by default "Preset # ").

Once the name of the new preset has been added to the list, it is necessary to press Ok or Apply in order for the audio preset to be stored.



The new audio preset will be created with the current audio configuration.



In order to check or modify an audio preset, just select the configuration of your choice from the list and press the Modify button[2]. The "Preset name" indicates the label of the configuration under your focus. Any changes along the "AUDIO INPUT", "AUDIO OUTPUT" or "AUDIO MISC" menus would be applicable to the selected preset as soon as the OK button is pressed. If instead the Cancel button has been selected then the original preset configuration is kept.

Keep in mind that the edition of any preset configuration will not disturb the current audio settings in operation.

Note: While the current audio configuration matches one of the stored Presets, this will be indicated on the second line of the Preset Configuration panel as "Current Preset: <…..>". Any mismatch between the Preset Configurations and the actual audio configuration forces the indication "Current Preset: <none>".

VI.3 Loading a preset

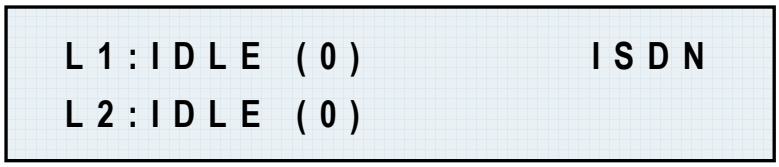
Once a preset has been selected, its configuration can be invoked as active or current by clicking on the "Load" button. The new configuration can be checked by moving through the different audio configuration menus. Until the "Apply" or "Ok" buttons are not pressed, the new audio configuration will not take effect.

[2] This option is available from version 5.4.0 onwards

Operation guide

VII.1 Starting the unit

Once the power switch is turned on, the starting sequence takes place. This sequence takes about 30 seconds, and as soon as it finishes, the display shows the main screen.



From the display it is possible to obtain information about the selected network, as well as the line status and decoder status. These are the different line status for PortaNet:

"IDLE" → The link line is available but it hasn't been made any connection.
"DOWN" → There is no link line plugged into the unit. The cable is not plugged in correctly or there is an fault on the line.

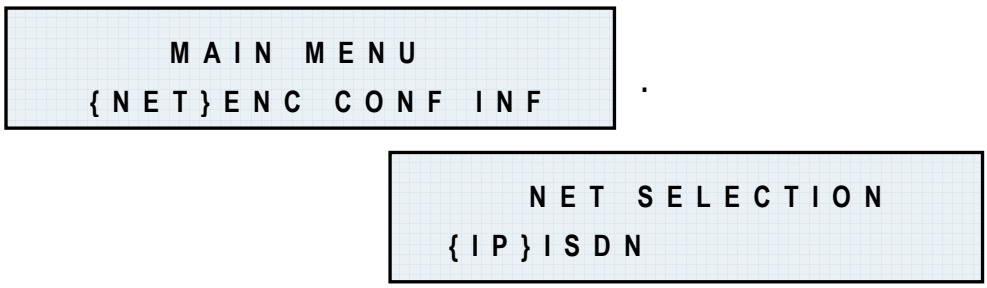
VII.2 Checking the configuration – the inf key

This key provides the user with a simple and quick way to display detailed information about the status and configuration of the PortaNet. As the screen is not able to display all the information at once, you can cycle through the different information screens by repeatedly pressing the INF button. These screens are:

1. Battery load information.
2. Audio input VU meters.
3. Audio output VU meters.
4. Decoding algorithm.
5. Encoding algorithm.
6. IP configuration parameters.
7. General configuration: NET type selected (IP, ISDN), audio input, etc.
8. GPIO status.

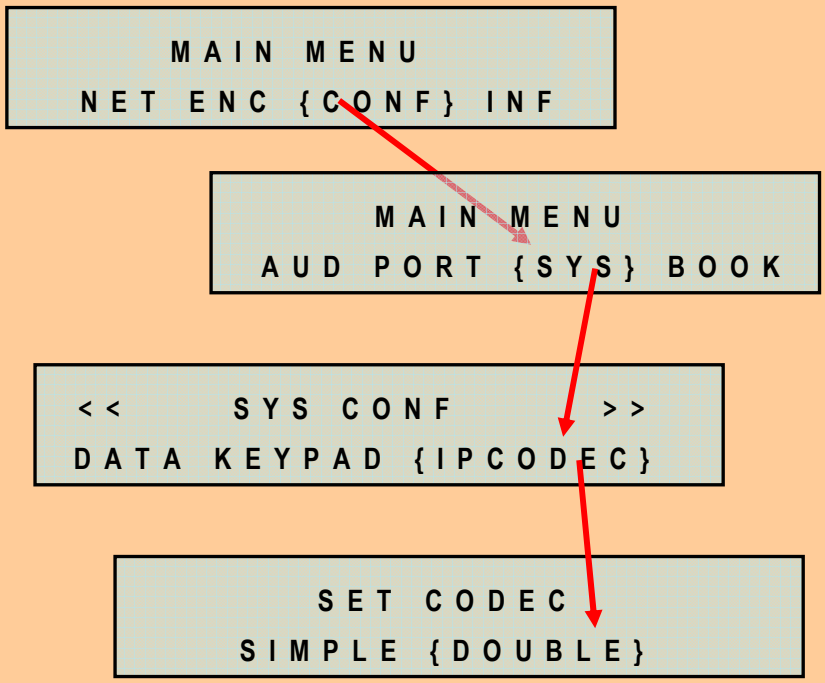
VII.3 Selecting the communication interface

PortaNet is able to communicate both via ISDN and via IP. The interface can be selected from the NET option within the menu.



IP can be used for remote control/monitoring of the unit even when NET = ISDN.

When IP network is selected, make sure that the unit has been configured to support two independent communications, that is, IPCODEC as "Codec Double":



'Simple' should be used only for stereo communications.

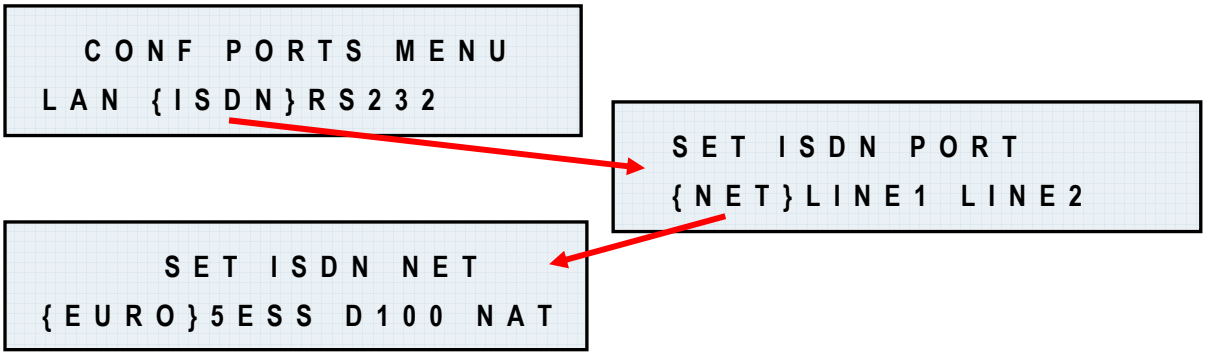
VII.4 Configuring the communication interface

Before making any connection it is advisable to check that all the communication interfaces are configured properly. The following is a quick guide to check this configuration and how to modify it when necessary.

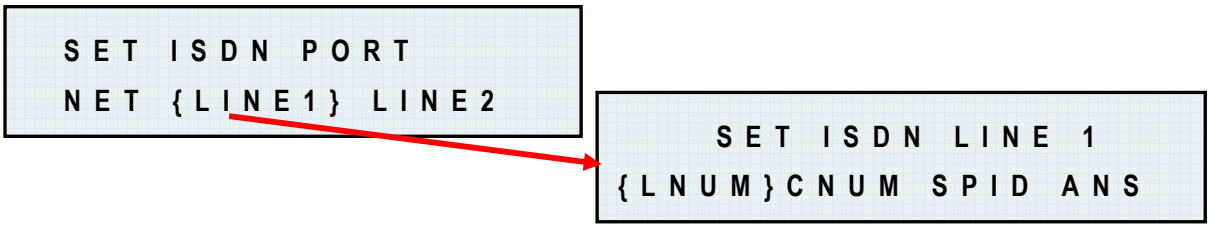
VII.4.1 Configuring the ISDN Terminal adapter

The ISDN Terminal adapter carries out all the signalling protocol in an ISDN communication. The parameters to configure it are the following:

a. **Communication protocol** → Check that the selected protocol is compatible with the line provided by your Telco. Most countries except the USA now use EISDN.



b. **Local number or 'LNUM'** → An abbreviation for 'Local Number'. With this option it is possible to assign a different number to each of the lines available on the basic rate ISDN. In this way we can automatically map incoming calls to each line, as each line will only respond to calls made to this local number. When the local number is programmed the unit only detects as an incoming call those calls that are made to the local number - all other calls are not recognised.



c. **CNUM** → An abbreviation for 'Calling Number'. This option allows you to program up to a maximum of three numbers that will work as incoming call filters. When the CNUM option is activated, the PortaNet will respond automatically only to calls made from these CNUM numbers. It is possible to program each line separately.

d. **SPID** → When you are using NAT1 or DMS100 you must enter the correct Service Profile ID (SPID) numbers for the ISDN BRI circuit.

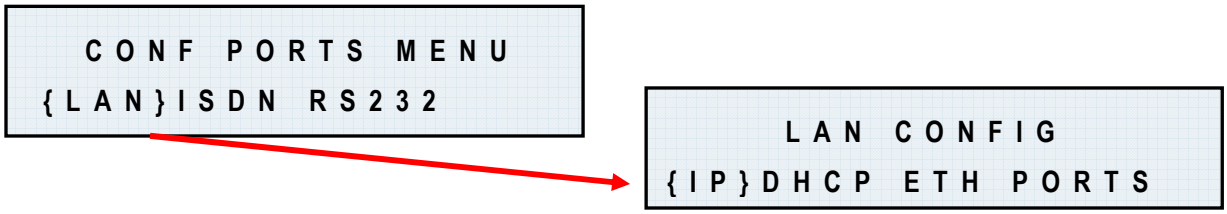
e. **ANS** → Automatic or manual response: When automatic response is selected, the unit will connect automatically to an incoming call, always doing so when there is no restriction imposed by the local number or other such call filters. In manual mode, the incoming calls are indicated on-screen and the user simply presses the corresponding CALL key to go ahead and connect that line.

VII.4.2 Configuring the LAN port

Before making a connection over any IP network it is necessary to configure the IP addressing parameters: IP address, netmask and gateway. PortaNet allows the assignment of the IP parameters either manually or by DHCP.

- Manually → User must enter the following IP parameters manually:

1. IP address.
2. Mask.
3. Gateway IP address



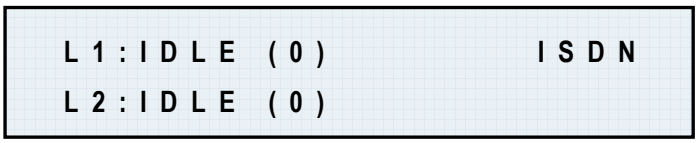
- Automatically → The unit gets the configuration automatically from a DHCP Server when starting.

VII.5 Checking the communication interfaces

Once the interface has been properly configured, the user can make some tests to confirm that the unit is ready to work. An easy way to check that everything is working fine is to make a 'loop' call, by calling our own unit.

VII.6 Connecting PortaNet to the line

Either using ISDN or IP, when plugging the link line cable into the unit, the display should indicate this with the word "IDLE" on the screen.



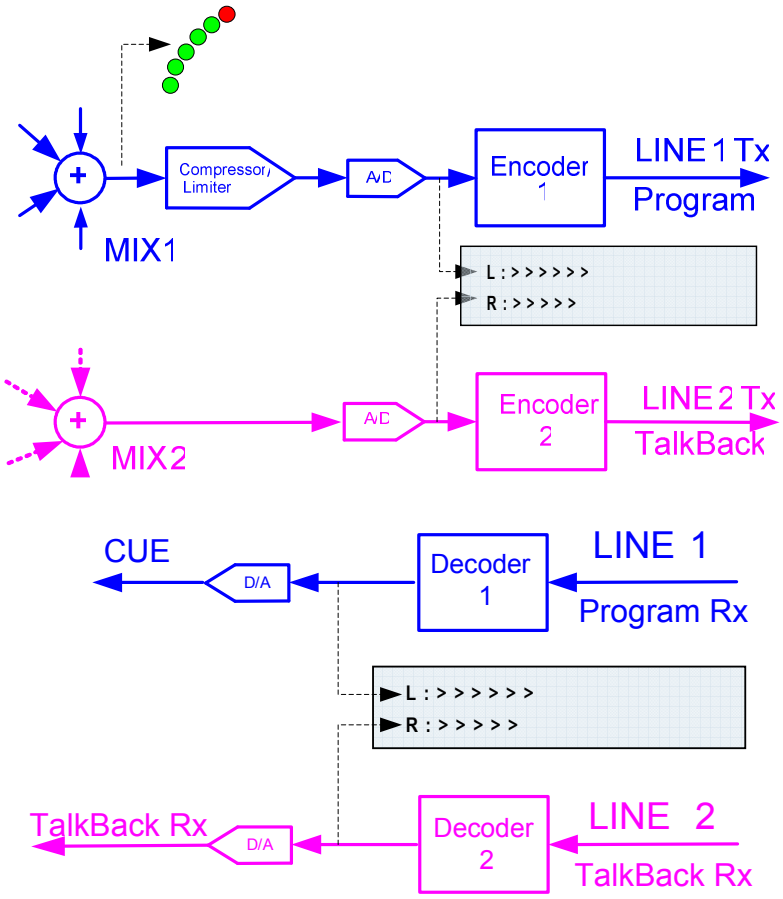
When no link line is detected on the communication interface, the word "DOWN" is displayed.

VII.7 Audio checking

PortaNet is equipped with a Vu meter on the top panel, and furthermore it offers the possibility to get audio level information from the display by pressing the INF key in the menu keypad. Let's take a look at both types of information:

The VU meter on the top panel measures the audio level just before the compressor/limiter. This level corresponds to the sum of all the inputs which contribute to the program line (line 1), that means, all those inputs whose led is on GREEN. This Vu meter comprises several green leds and a red overload led.

The audio levels obtained from the display with the INF key are digital domain measurements. The input levels are measured before the compression and the output levels are obtained after de decoding process, as can be seen from the following diagrams:

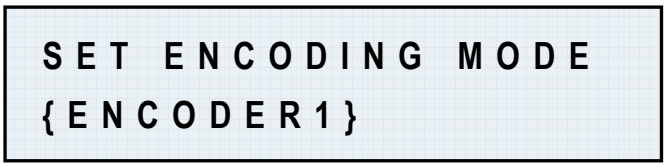


VII.8 Selecting the compression algorithm

The ENC option in the menu allows the user to change the encoding mode. The following points must be taken into account when configuring the encoder:

1. The encoder of line 2 or TalkBack line cannot be configured in any mode other than G722 when IP is the selected network. Over ISDN there are no restrictions regarding the available compression modes for this encoder.
2. The AUTO mode causes the encoder to adopt the configuration detected by the corresponding decoder.
3. The available options will depend on the selected network (IP or ISDN) due to some crucial differences such as the line bandwidth.

When selecting the ENC option on the main menu the following screen appears:



The different options for the Encoder 1 are as follows:

- AUT : Automatic mode.
- PCM : Linear audio.
- G711.
- G722.
- MPL2 : MPEG 1,2 Layer II.
- MPL3 : MPEG 1,2 Layer III.
- AAC2LC: AAC MPEG 2 Low Complexity.
- AAC4LC: AAC MPEG 4 Low Complexity.
- AAC4LD: AAC MPEG 4 Low Delay.
- AAC4HE: AAC MPEG 4 High Efficiency.
- Apt-X^{TM} : Standard and Enhanced

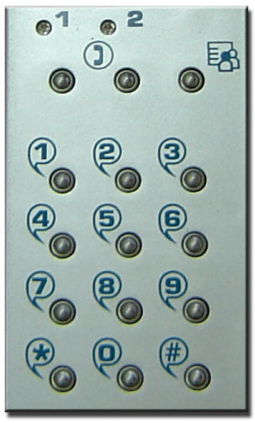
VII.9 Decoder operation

The Decoder is totally automatic, except when working over ISDN and receiving J52 and aptX$^{TM.}$
The Decoder of line 2 or TalkBack line is fixed in G722 when the IP interface is selected.

VII.10 Calling from PortaNet

PortaNet can establish connections both via ISDN and IP networks. The procedure is very similar and can be carried out from the web page or the main menu. Let's see how to establish a call from the top surface control keypad.

VII.10.1 CALL1, CALL2 and Phone Book keys



The CALL1 and CALL2 keys give the user access to the call control menu for line 1 and line 2 respectively. The Phone Book key is used for establishing communications using a configuration saved in the Address Book. This book can store both ISDN and IP numbers. Also, each entry of the book can store the compression mode to be used when connecting to the stored number. It is worth mentioning that the encoder of line 2 or TalkBack line cannot be configured when working over IP, and that it is fixed in G722 mode and any other algorithm is discarded.

The fundamental role of these keys is the initiation and termination of a connection. Note that the keys have different functions depending on the communication line status at any given moment. Additionally, the display will also show different information depending on the type of NET selected - IP, or ISDN.

Depending on the line's status, CALL 1 and CALL 2 will function in the following way:

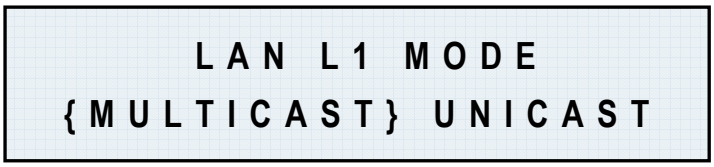
- With the line disconnected:
  a) Initiation of a communication: Pressing CALL 1 or CALL 2 will access the call initiation menu. Depending on the type of NET selected (IP or ISDN) the dialog that you will see will vary. (Note that, CALL 2 only works when PortaNet works as a DUAL codec[3]).
  b) Connecting to an incoming call: if the PortaNet is pre-configured to manual answering, the CALL 1 or CALL 2 keys will connect the line when an incoming call is detected on the respective lines.
- With a line connected: There are two different functions:
  a) A short press will display on-screen the number or IP address of the current connection.
  b) A long press (over 1 second) will disconnect the associated line.

It is possible to program a number of call retries from the PortaNet web page.

[3] DUAL codec: PortaNet is able to operate as a DUAL codec over ISDN and over IP (when configured in 'DOUBLE' mode).

VII.10.1.1 Establishing a call when PortaNet is configured as an IP Unicast/Multicast codec

Pressing CALL 1 the screen will display the following:

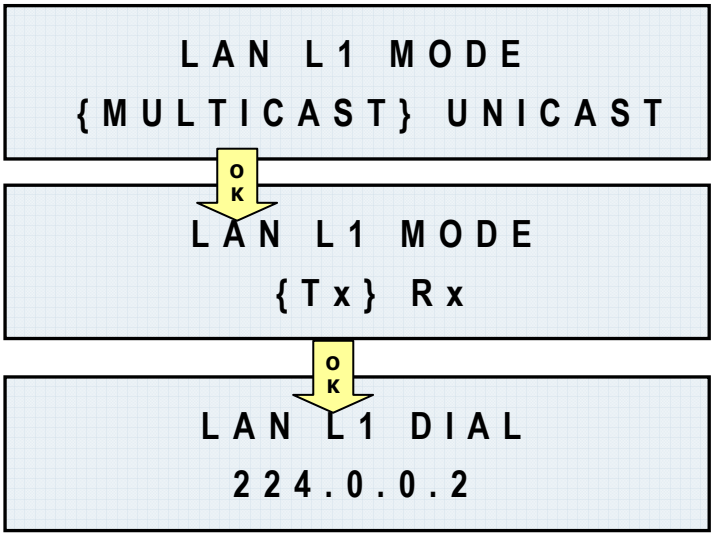


The user must select which kind of communication will be used:

- Unicast Bidirectional → This is a bidirectional point to point connection, that is, both ends will transmit and receive audio simultaneously. It will be necessary to check the upload and download bandwidth on the link.
- Unicast Unidirectional Tx → This is a unidirectional point to point communication where only the end which makes the call will send audio.
- Unicast Unidirectional Rx → This is a unidirectional point to point communication, where the calling end will be the receiver.
- Multicast Tx → This is point to multipoint communication in which the calling end will join a multicast group as a transmitter.
- Multicast Rx → This is point to multipoint communication in which the calling end will join a multicast group as a receiver.

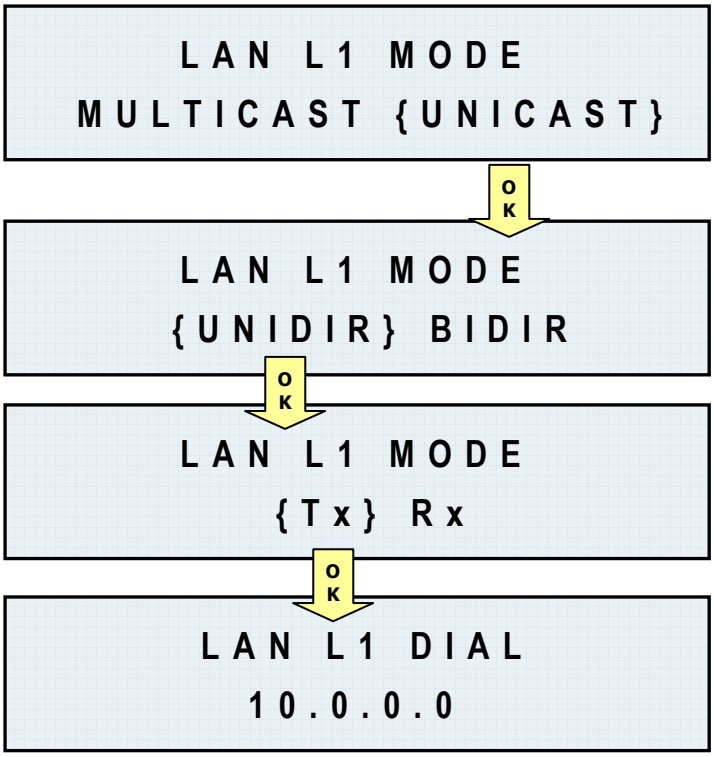
Next, depending on the selected option will show:

MULTICAST →



To make the connection press OK. "Period" is selected with the # key.

UNICAST →



To make the connection press OK. "Period" is selected with the # key.

VII.10.1.2 Establishing a call when PortaNet is configured as an IP Multi-Unicast[4] codec (NET = IP)

Pressing CALL 1 the screen will display the following:

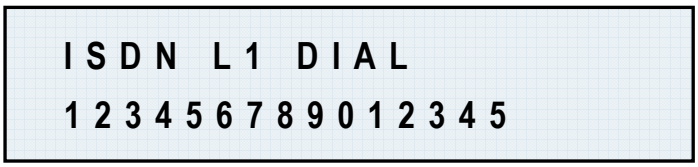
Group Redial is the only option available when calling from the front panel. This option will try to re-establish all previous connections, both Tx and Rx.

CALL 2 key is not available under Multi-Unicast mode.

[4] This option is available from version 5.2.1 onwards.

VII.10.1.3 Establishing a call when PortaNet is configured as an ISDN codec (NET = ISDN)

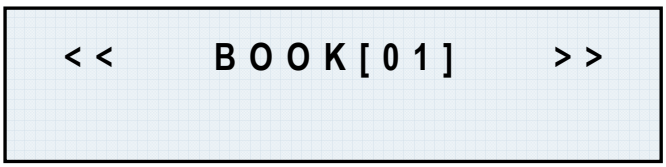
Pressing the CALL 1 key will display the page where you can enter the number to be called on Line 1. The CALL 2 key does the same for Line 2.



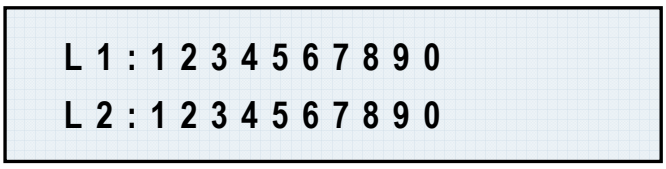
VII.11 Calling from the Phone Book

This key directly accesses the internal Address Book. By pressing this key we can view and connect to entries in the Address Book (NOTE: To edit or modify entries go to the BOOK option in the main menu). For making connections however, the BOOK key works like this:

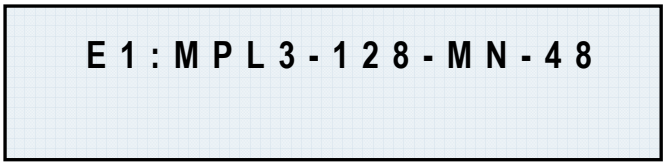
Pressing BOOK you enter the Address Book index:



Use the left and right navigation keys to go through the different entries in the Address Book index. By pressing OK we select the item that is on-screen. Now the screen shows the number/numbers to call, or the IP addresses, depending on which ones have been saved.



Pressing OK again will process this information, that is, the PortaNet will configure the Encoder according to the BOOK entry and make the connection over ISDN or IP. By pressing the right arrow key instead of the OK one, the unit will display the saved encoding mode.



When you press the BOOK key you will only be shown those entries that are relevant to the NET that is currently activated. So when NET = ISDN, you will only be able to browse previously stored ISDN entries, just as when NET = IP you will only be able to select from relevant IP addresses.

VII.12 Incoming calls

The incoming calls can be connected on line 1 or line 2. It is worth mentioning here that each line has a different application and, for example, over IP, there are different configuration options (line 2 only encodes G722). Line 1 will be the main program and line 2 the coordination or TalkBack program.

Next follows some recommendations to avoid confusion when controlling the lines:

VII.12.1 Receiving ISDN calls

Compatibility: it is compatible with any ISDN audio codec with the same compression modes.

Compression algorithms: There are no restrictions regarding the available compression modes in any line. Encoder and decoder can work in different modes independently.

Automatic and manual answer: Incoming calls can be answered automatically or manually depending on the menu option 'ANS'. If the 'ANS' mode is set to manual, the corresponding CALL key must be pressed to accept the call and connect to it.

- Automatic answer → The call is picked up automatically.
- Manual answer → In this mode, the unit will warn about the incoming call by blinking the LED of the corresponding line. To pick up the call, press the corresponding CALL key.

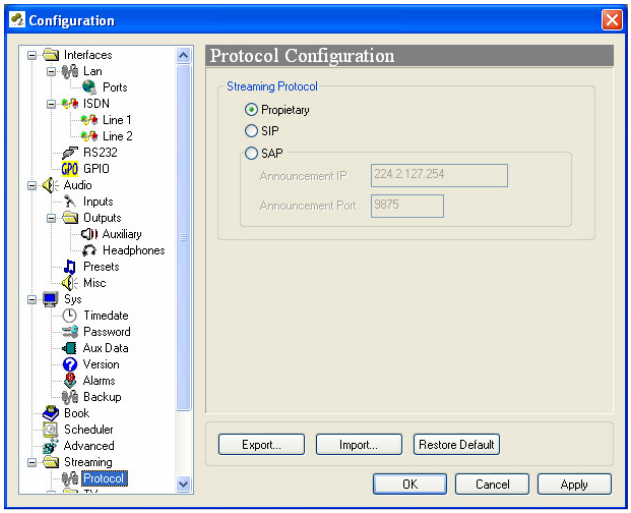
Mapping Incoming calls: With the LNUM (local number) option it is possible to assign a different number to each of the lines available on the basic rate ISDN interface. In this way we can automatically map incoming calls to each line, and each line will only respond to those calls which were made to this local number. When the local number is programmed the unit only detects as an incoming call

those calls that are made to the local number - all other calls are discarded. Thus, the user can control which calls will be connected on program line (line 1), and which ones on the coordination line (line 2).

When there is only one number available for both channels, the first call will be connected on line 1 and the second one on line 2. Thus, before making connections for establishing the TalkBack channel, make sure that the program line has been already connected.

VII.12.2 Incoming calls via IP

Compatibility: PortaNet supports a proprietary IP streaming protocol, only compatible with Prodys IP audio codecs. In addition, support for industry-standard algorithms is provided under the 'Streaming Protocol' tab in the Configuration window of PortaNet web page. A preliminary version of all protocols defined as 'mandatory' by the EBU group in their N/ACIP (Audio Contribution Over IP) project is supported. This project defines the standard to achieve interoperability between audio codecs over IP.



Available compression modes: Line two can only be configured to G722 (encoder and decoder). Line 1 can be configured to any compression mode. The encoder and the decoder of line 1 can work in different modes.

Automatic answer: The answer mode is always automatic, so the call is answered automatically.

Incoming call mapping: it is not possible to map the incoming calls to a line. And the following rules apply:

- All the calls made from line 1 in ProntoNet or PortaNet will be received through line 1.
- All the calls made from line 2 in ProntoNet or PortaNet will be received through line 2.

PortaNet remote control

PortaNet can be controlled remotely by using an Internet Explorer web browser connected through the LAN port. The computer can be locally connected directly via a crossover CAT-5 cable, or remotely from a computer connected to the LAN. To access the PortaNet using Internet Explorer, enter the IP address of the unit in the address bar. Keep in mind that the PortaNet factory IP address is 192.168.100.100 and it could be necessary to modify the network configuration of the computer on which the web browser is running.

Installation Requirements

1.- Pentium 166 or higher.

2.- 64MB RAM minimum.

3.- Operating Systems:

Microsoft Windows XP, Microsoft Windows 2000,

Microsoft Windows NT 4.0 Service Pack 6 or higher,

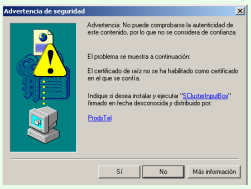
Microsoft Windows Millennium Edition (ME), Microsoft Windows 98.

Microsoft Windows Vista.

4.- Microsoft Internet Explorer 5.0 or higher.

The screen resolution must be 1024x768 minimum.

The first time that the computer accesses the PortaNet it is necessary to install the software. The computer will show the following window:

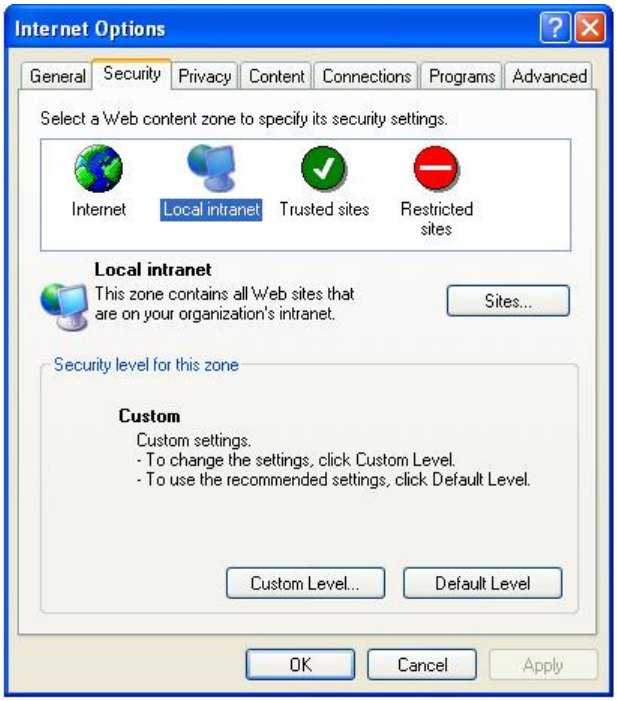

Advertencia de seguridad
Advertencia: No puede comprobarse la autenticidad de este contenido, por lo que no se considera de confianza.
El problema se muestra a continuación:
El certificado de raíz no se ha habilitado como certificado en el que se confía.
Indique si desea instalar y ejecutar "SClusterInputBox" firmado en fecha desconocida y distribuido por:
ProdaTel
Sí
No
Más información

The unit is supplied with the following IP address: 192.168.100.100

The first time the user accesses the PortaNet web page, an OCX file has to be downloaded and installed on the computer. Microsoft Internet Explorer can be configured to block OCX objects installation and/or execution. So, depending on the configuration of the web browser, a message like the following might appear when first accessing the PortaNet web page:

Internet Explorer - Security Warning
Windows has blocked this software because it can't verify the publisher.
Name: SClusterOutputBox.cab
Publisher: Unknown Publisher
OK
To help protect your computer, Windows will block software when it cannot verify the publisher. See how this helps protect your computer.

Go to Internet Options in IExplorer, click on 'Security' tab, and set 'prompt' when downloading ActiveX signed and unsigned controls at Local and Internet zones.

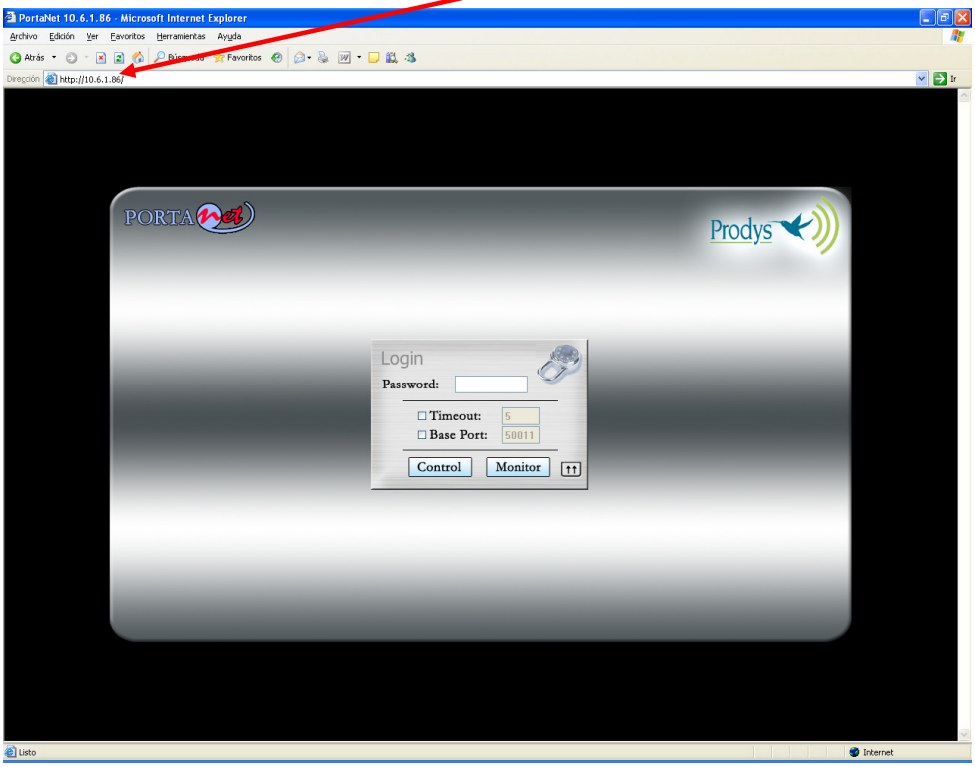


Each firmware version might have a different OCX file, so the new OCX should be installed as it is done for the first access to the web page of the unit. If the unit was upgraded and, depending on the 'cache' configuration of the Internet explorer, there might be problems when accessing the web page, given that the old web page might be offered by the browser instead of the real one, which should be installed to replace the old one. In this case, a message indicating 'Incorrect Versions' will appear as soon as the user click on 'Control' or 'Monitor' on the login page. Click on F5 to skip the cache entries, and access to the 'real' web page. Even after pressing F5 and, depending on the IExplrorer configuration and/or version, this situation might continue. In that case, go to Internet Options in IExplorer, click on 'General' tab, and delete temporary files

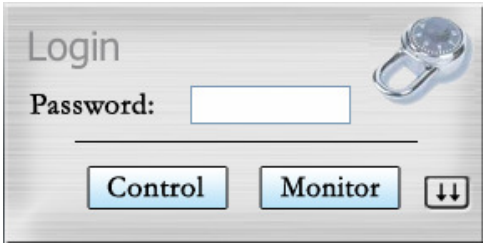
Windows Vista: Should the user experience a problem when downloading the OCX file when first accessing the web page of the unit, please disable UAC (User Access Control) on Windows Vista. Once the OCX file has been installed in the computer, UAC can be enabled again.

VIII.1 Getting Started

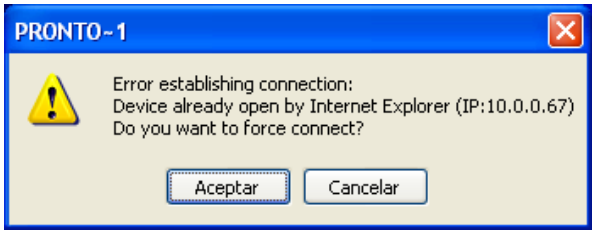
To access the PortaNet from the Internet Explorer enter the IP address of the unit in the address bar as shown here:



The user can choose whether to monitor or to control PortaNet from the Web Page. Bear in mind that only one page at the same time can control the unit. However, it is possible to monitor the unit from several web browsers simultaneously.

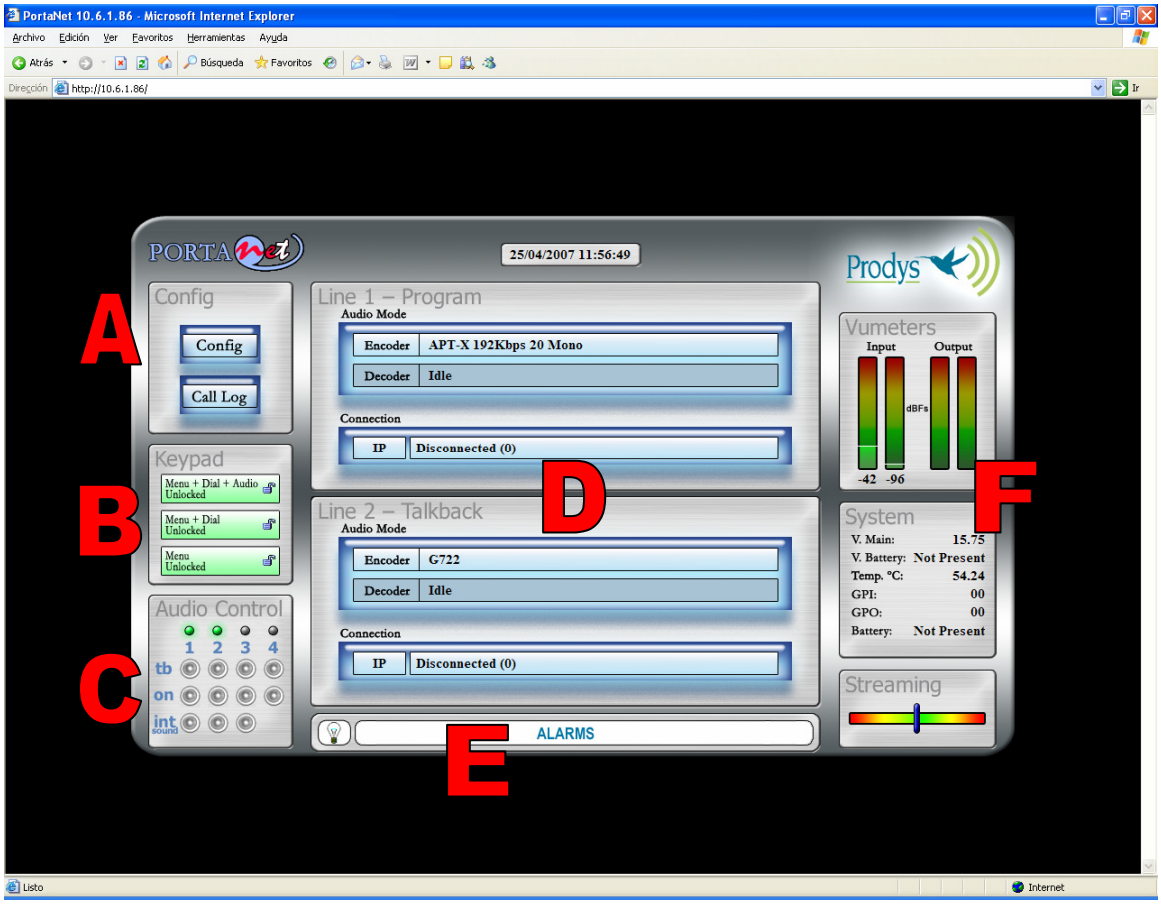


If a unit is already being controlled by a web page and we try to get the control from another website, a message will appear. This message will indicate that the unit is already being controlled from another PC and the IP address of this computer.



It is possible to get the control by pressing OK. Then, the connection of the current controller will be closed and the unit will be available for the new controller.

Once the password is entered correctly, the web browser will display the "Home Page":

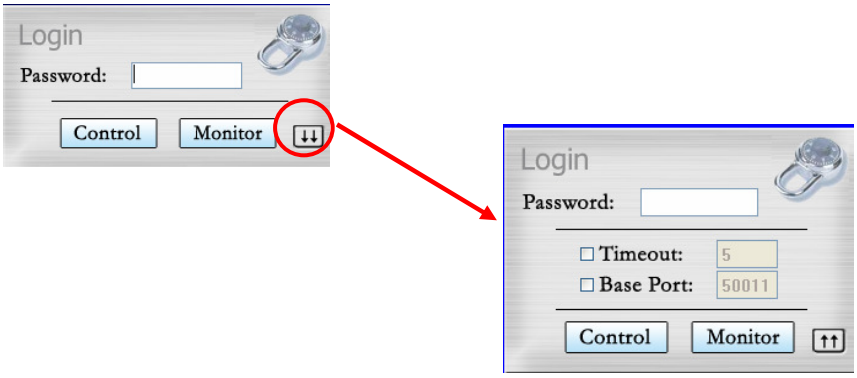


The Web page is arranged in the following main areas:

- **A.** General Configuration area.
- **B.** Keypad locking control.
- **C.** Audio keypad remote control.
- **D.** Audio encoding configuration and communication control.
- **E.** Alarms area
- **F.** Monitor area.

VIII.1.1 Extra Options in the Login Dialog Box

By clicking on the right down corner of the login dialog the screen will show two more options:

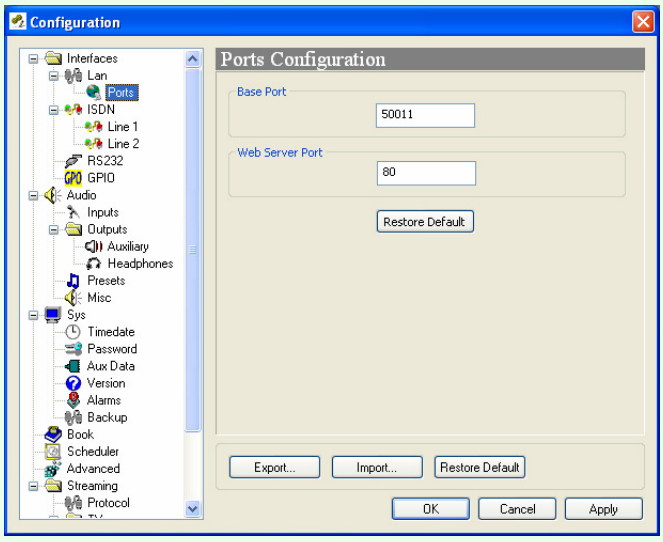


- **Timeout:** If an acknowledgment has not occurred in a specified amount of time, the timeout ends the waiting loop. In some network conditions could be necessary to increase the time out in order to get the PortaNet control.

- **Base Port**: When PortaNet ports have been modified, it will be necessary to change this parameter according to the new port configuration in order to get to access to the PortaNet web browser.

MORE ABOUT THE PORTANET PORTS

Changing Portanet Ports: The PortaNet configuration menu allows the user to configure which ports the unit will use for its TCP/UDP/IP communications from the web page.



There are two different groups:

- **Web Server Port**: By default, it is TCP port 80. This is the internal web server port.

- **Base Port**: By default, it is 50011 for TCP and UDP ports. This is the first port of the range of ports used by the unit. From this base port on, up to 30 ports should opened/forwarded. That is, if the base port is set to 50011, the range of ports goes from 50011 to 50041, both for UDP and TCP, should be opened/forwarded in the corresponding router/firewall (when required).

IMPORTANT

The following should be taken into consideration when changing the web port:

- **To access the web page of the unit, the new port has to be indicated in the http address bar of the web browser after the IP address, separated by a colon:**

  http://<IP>:<Port> Example: 192.168.0.10:8080

VIII.2 General configuration

By clicking on the 'Conf' key the user can access the general configuration window. The different options are arranged in the following way:

- **Interfaces**: To configure the PortaNet Ports.
- **Audio**: Different options regarding audio configuration can be set from this menu: remote audio input level control, headphone output configuration or audio presets for example.
- **Sys**: System configuration: clock, keypad locking...
- **Book**: To edit the phone book.
- **Scheduller**: To schedule calls.
- **Advanced**: Special options.
- **Streaming**: To configure IP related parameters and to test the quality of the IP connection between units.

VIII.2.1 Interfaces

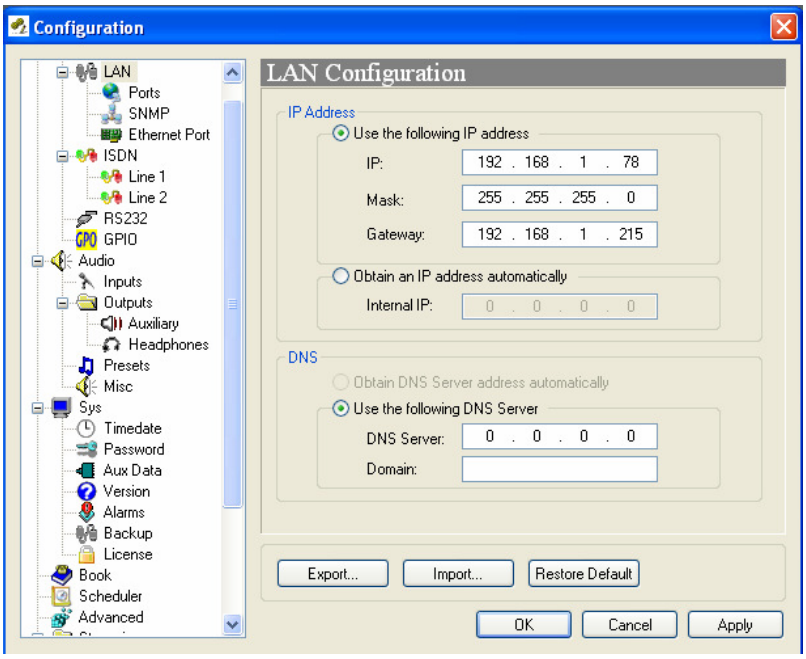
By clicking on the **Interfaces** icon the port configuration dialog appears. In the left side the ports can be selected. The right window shows the window dialog to configure the selected port. For example, if we select the LAN interface:

VIII.2.1.1 LAN port

PortaNet allows the assigning of the IP parameters both manually and automatically (DHCP).

- Manually → User must enter the following IP parameters manually:

4. IP address.
5. Mask.
6. Gateway IP address

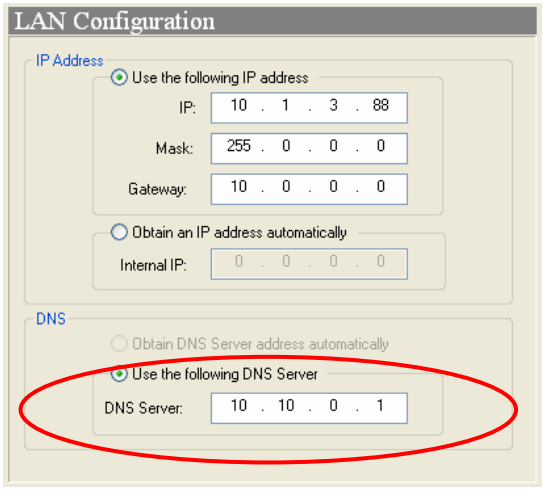


- Automatically → Check "Obtain an IP address automatically" option to get the configuration automatically from a DHCP Server.

By setting DHCP, the unit will receive its IP parameters when starting. These IP settings might be different from time to time, that is why PortaNet supports RIP2 protocol. This protocol allows the user to set an 'internal' IP address, in order that the unit can be identified regardless of the IP settings provided by the DHCP server.

- **DNS support:**

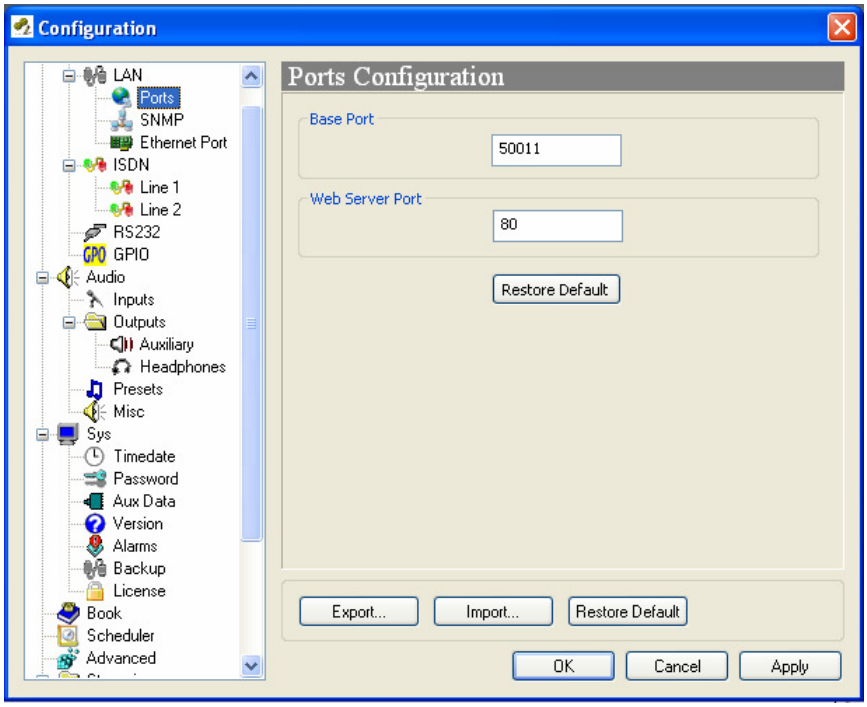
By specifying the IP address of a DNS Server, the user can make calls to 'names' instead of IP addresses. This new feature is available in any place where an IP address has to be entered. For example: the dial window, the phone book, SNTP IP server address…This is the DNS configuration windows:


LAN Configuration
IP Address
Use the following IP address
IP: 10 . 1 . 3 . 88
Mask: 255 . 0 . 0 . 0
Gateway: 10 . 0 . 0 . 0
Obtain an IP address automatically
Internal IP: 0 . 0 . 0 . 0
DNS
Obtain DNS Server address automatically
Use the following DNS Server
DNS Server: 10 . 10 . 0 . 1

As can be seen from the picture, the DNS information can be obtained from a DHCP Server.

VIII.2.1.1.1 Ports

It is possible to configure which ports the unit will use when selecting Prodys proprietary protocols for IP communications[5].



[5] For more information, please read chapter IX.6.6– Prodys Proprietary Protocols

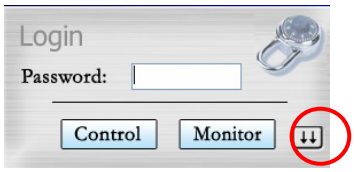
There are two different groups:

1. **Web Server Port**: By default, it is TCP port 80. This is the internal web server port.
2. **Base Port**: By default, it is 50011 for TCP and UDP ports. This is the first port of the range of ports used by the unit. From this base port, up to 30 ports should opened/forwarded. That is, if the base port is set to 50011, the range of ports goes from 50011 to 50041, both for UDP and TCP, should be opened/forwarded in the corresponding router/firewall (when required).

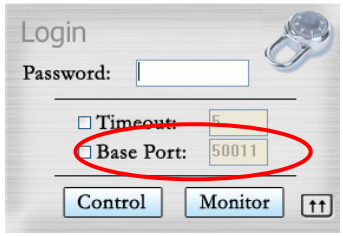
The following things should be taken into consideration when changing the base port:

1. To access the web page of the unit, the new port has to be indicated in the http address bar of the web browser after the IP address, separated by a colon:
   http://<IP>:<Web Port> Example: 192.168.0.10:8080

2. To log into the unit, click on the advance features button of the login window:

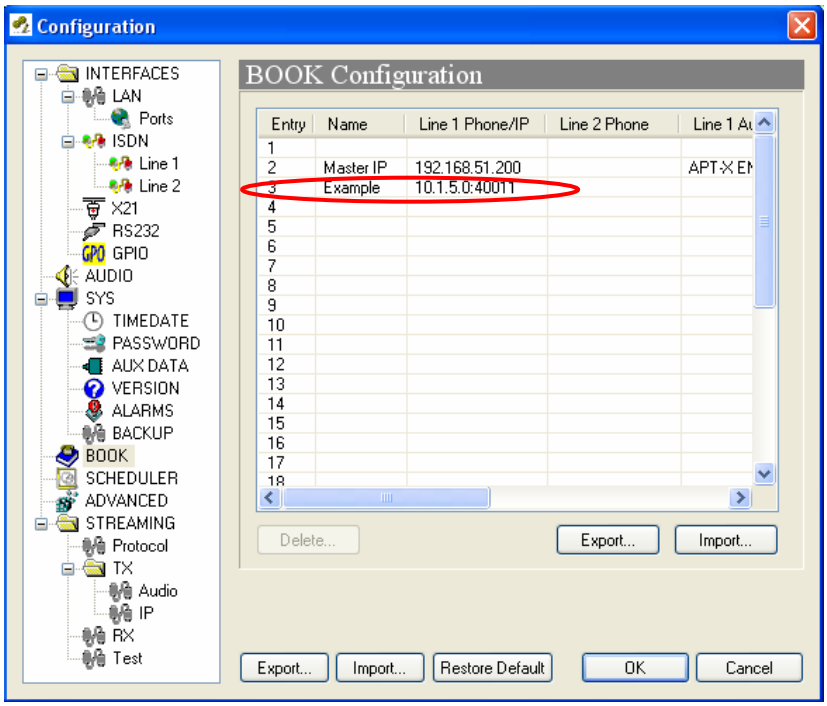


   and specify the base port which we want to connect to:



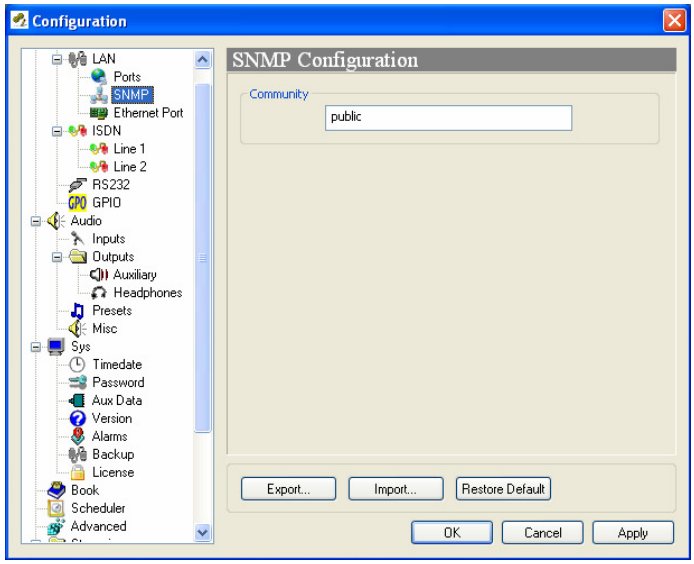
3. When establishing a call, the new base port must be entered after the IP address, separated by a colon. Example: When calling to a unit which IP address is 192.168.1.2, and base port 40011, the following call destination should be entered into the 'Dial' window: **192.168.1.2:40011**.
4. When establishing a bidirectional call with Prodys IP codecs, two connections are made automatically, one for each direction. For the

receiver to call properly to the caller (*only in case the base port in the caller is not the default one*), it should have an entry on its phone book with the IP address of the caller and the base port of the caller. Example: if the caller has the IP address 10.1.5.0 and its base port is 40011, in the receiver, the following entry should be configured in its phone book:



VIII.2.1.1.2 SNMP configuration[6]

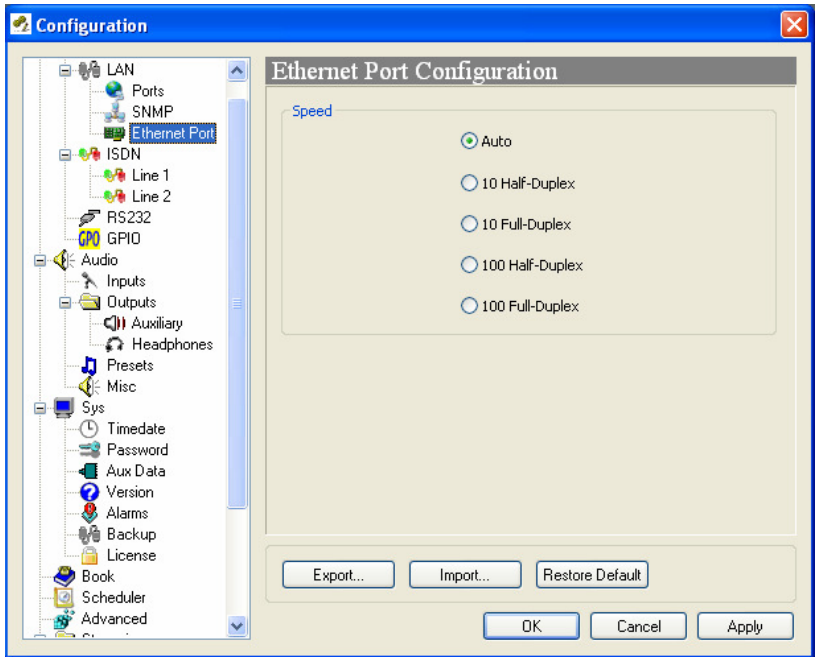
It is possible to set the SNMP community.



[6] This option is available from version 5.2.1 onwards.

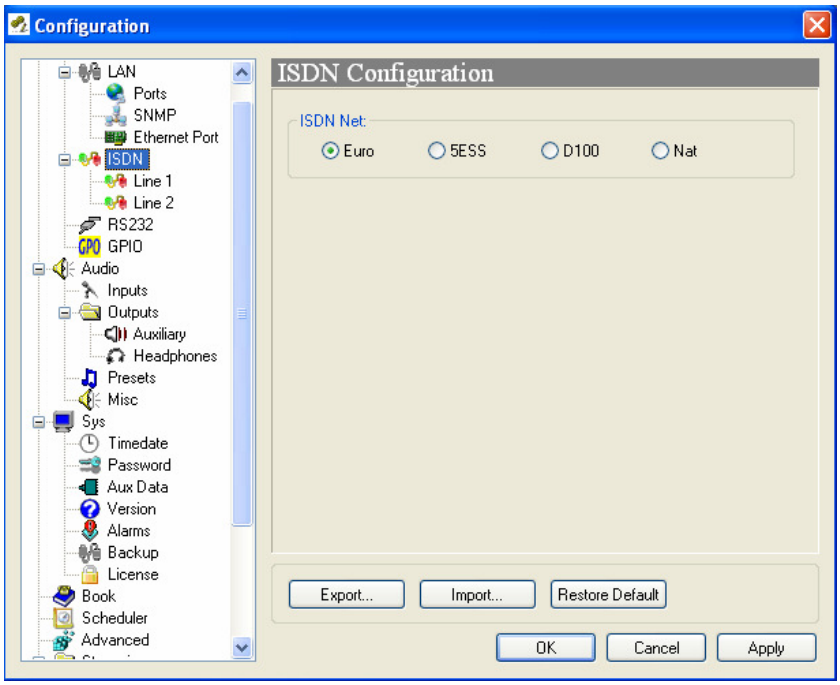
VIII.2.1.1.3 Ethernet Port Configuration

It is possible to set speed and duplex configuration to the following values:

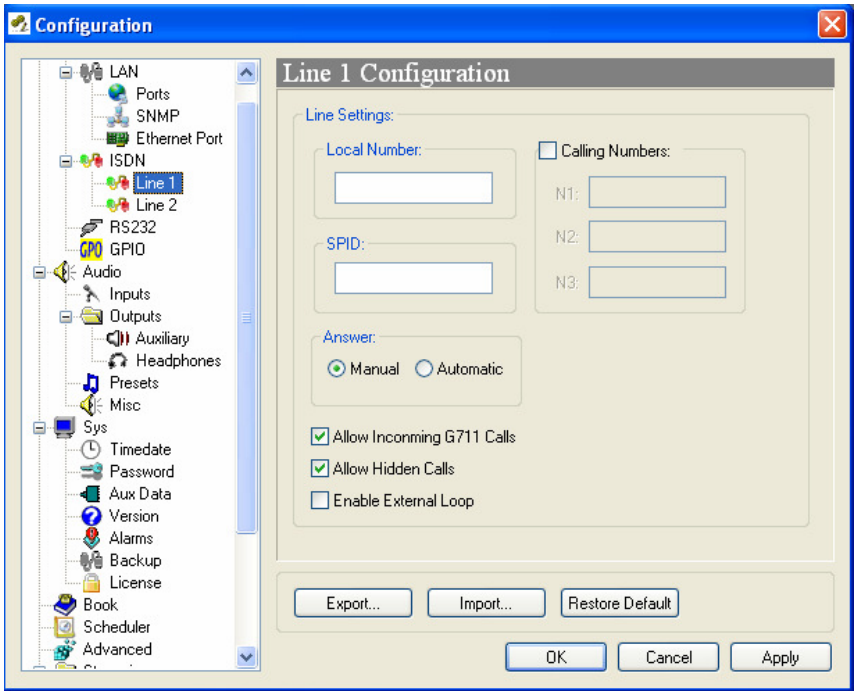


VIII.2.1.2 ISDN Terminal adaptor Configuration

The first dialog allows the selection of the ISDN protocol. There are four options: Euro-ISDN, 5ESS, DMS 100 and NAT1:



The options Line 1 and Line 2 allows the user to configure in each line (B channel) the Local Number, Calling number filters and the answering mode:



- **Automatic or manual response:** Incoming calls can be answered automatically or manually depending how the menu option ANS is configured for the ISDN port. If the ANS mode is set to manual, the appropriate CALL key must be pressed to accept the call and connect to it.

- **Call filters:** It is possible to record up to three numbers for each line that work as call filters, meaning that the line will only connect to calls that come from these pre-programmed numbers. This option is found in the ISDN set up menus under CNUM (Calling Number).

- **Local number**: It is also possible to assign a single number to each line in a way that the line will only respond to calls to this local number. This can be used if you need to map an ISDN directory number to a specific audio port. This option is found in the ISDN set up menus under LNUM (Local Number).

  - Enabling/disabling G711 ISDN incoming calls
    This check box allows the user to enable/disable voice calls (G711) over ISDN.
  - Enabling/Disabling hidden-number ISDN calls

This check box allows the user to enable/reject incoming ISDN calls with undefined number.

- External ISDN loop[7].
  It is possible to set a loop for the ISDN interface, so that data coming from any ISDN connection will be sent back. This functionality works over both B channels.

VIII.2.1.3 RS232 Port

There is one RS232 ports for use as auxiliary data ports. This port allows the transmission and reception of data along with encoded audio. This port is always set to 8 DATA bits, NO parity, 1 START bit and 1 STOP bit. The bit rate can be adjusted to between 300 and 9600 bps via software.

VIII.2.1.4 GPIO Port

From this option the inputs and outputs of the GPIO port are configured.

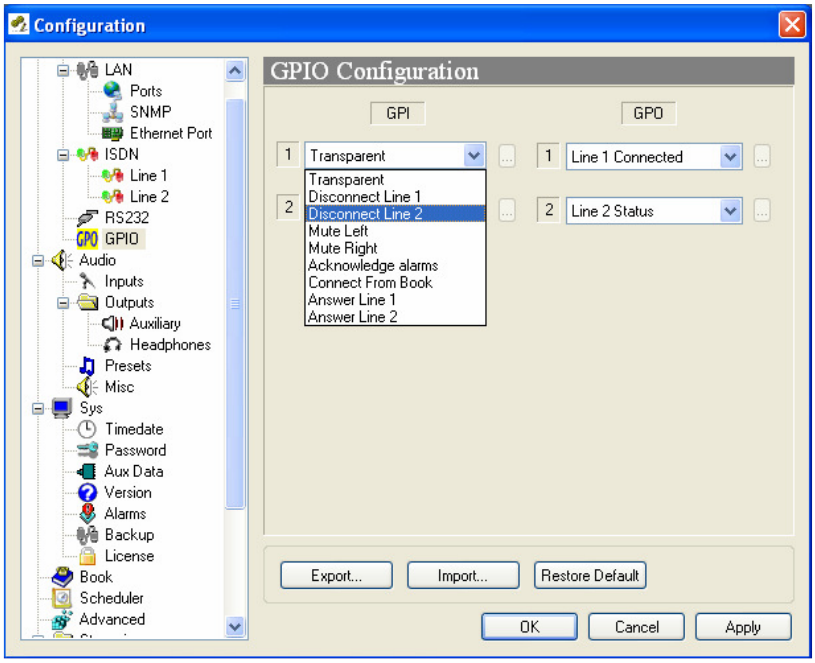
INPUTS	
Transparent	Under this configuration, the state of the input will be present in its homologous output in the PortaNet connected on the other end.
Connect From Book	When this input is activated, the PortaNet will proceed automatically to connect the line 1 or line 2 (as configured). The user can select which entry of the book will be used for this purpose[8].
Disconnect Line 1	When this input is activated, the PortaNet will proceed automatically to disconnect the line 1.
Disconnect Line 2	When this input is activated, the PortaNet will proceed automatically to disconnect the line 2. It is only useful when the PortaNet is working in ISDN mode as a DUAL CODEC.
Mute Left	When this input is activated, the left audio output will be muted.
Mute Right	When this input is activated, the right audio output will be muted.
Acknowledge alarms	When this input is activated, the alarms will be acknowledged.
Answer Line1[9]	Line 1 will be picked up when receiving a call.
Answer Line2	Line 2 will be picked up when receiving a call.

- **The inputs are activated by grounding**
- **Transparent configuration only works with audio algorithms that support ancillary data and have been previously activated.**

[7] This option is available from version 5.2.1 onwards.

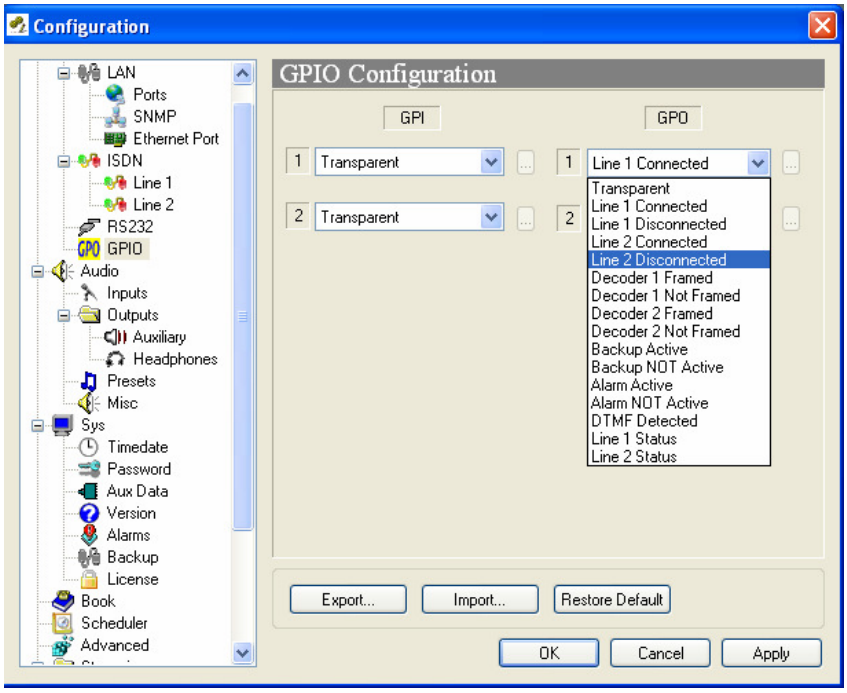
[8] Before version 5.0.0, only entries 1 and 2 of the phone book were used for calling line 1 and line 2.

[9] From version 5.2.1 onwards.

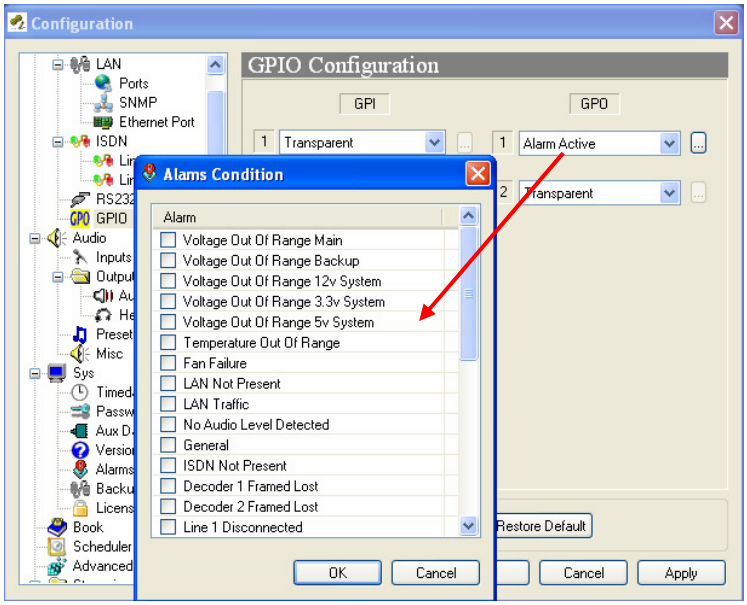


OUTPUTS	
Transparent	Under this configuration, the state of the output will be the same as the corresponding input of the Suprima connected on the other end.
Line 1 Connected	The output will be activated when the line 1 is connected.
Line 1 Disconnected	The output will be activated when the line 1 is disconnected.
Line 2 Connected	The output will be activated when the line 2 is connected. It is only useful when the PortaNet is working in ISDN mode as a DUAL CODEC.
Line 2 Disconnected	The output will be activated when the line 2 is disconnected. It is only useful when the PortaNet is working in ISDN mode as a DUAL CODEC.
Decoder 1 Framed	The output will be activated when the Decoder 1 is Framed.
Decoder 1 NOT framed	The output will be activated when the Decoder 1 is NOT Framed.
Decoder 2 Framed	The output will be activated when the Decoder 2 is Framed. It is only useful when the PortaNet is working in ISDN mode as a DUAL CODEC.
Decoder 2 NOT framed	The output will be activated when the Decoder 2 is NOT Framed. It is only useful when the PortaNet is working in ISDN mode as a DUAL CODEC.
Backup Active	The output will be activated when the ISDN Backup is working.
Backup NOT Active	The output will be activated when the ISDN Backup is NOT working.
Alarm Active	The output will be activated when one Alarm is activated.
Alarm NOT Active	The output will be activated when there are NOT Alarms activated.
DTMF detected	DTMF is detected by the Decoder.
Line 1/2 Status	Line Disconnected: GPO set to '0'. Line Connected: GPO set to '1'. Trying/receiving a call: GPO blinking (1 second period).

- **Transparent configuration only works with audio algorithms that support ancillary data and have been previously activated.**

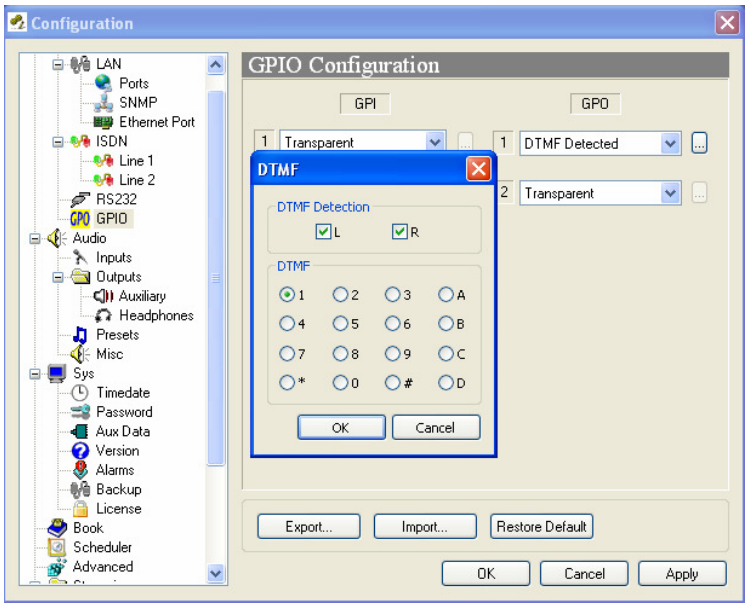


When a GPO is configured to monitor alarms ("Alarm active"), it is possible to select which alarm will enable this output.



This GPO will be activated when one or more of the selected alarms arise[10].

In addition, it is possible to program a GPO to be enabled when a DTMF is detected in either Decoder. The decoder number and the DTMF which will enable the GPO must be defined by the user.



VIII.2.2 System Configuration

Grouped here are functions that affect the general operation of the unit.

- **Loop**: This sets up an Encoder-Decoder loop. Its purpose is to help the user to find problems with audio connections. The loop takes place in accordance with the configuration available at that moment.
- **PLL**: This is a special option that allows you to select the PLL reference clock when NET = IP.

- **Redial**:It is possible to configure a number of redials. The PortaNet will try the connection in the following cases:
  1.- The unit is trying the connection for the first time.
  2.- The unit was connected and lost the connection because the line dropped or the line was disconnected from the other end.

- **Keypad**: This option allows the user to lock automatically the frontal panel keypad. Automatic means that the keypad is locked a short while (around 1 minute) after you just press a key. It is possible to unlock from the keypad by pressing Ok and then *.

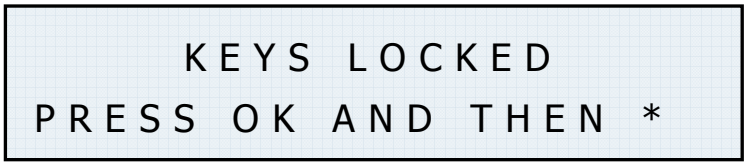
[10] This option is available from version 3.3.1.

It is also possible to lock the keypad with a password.

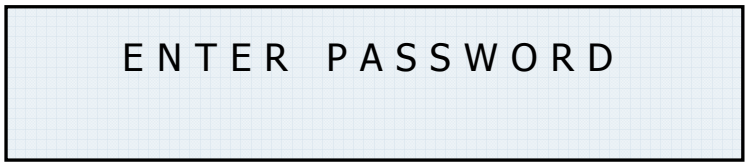
INF won't be disabled.

CALL1 and CALL2 keys will be available to answer any incoming call.

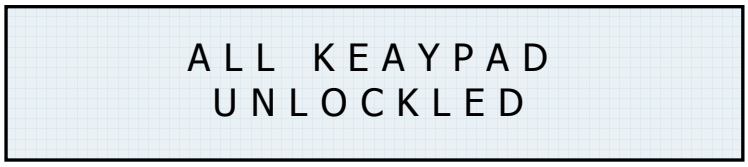
When the keypad is locked, and any key is pressed, the following message appears:



If password protection was enabled, the user will be prompted to enter the password, and the following message appears:



If the password is entered successfully, the following message appears:

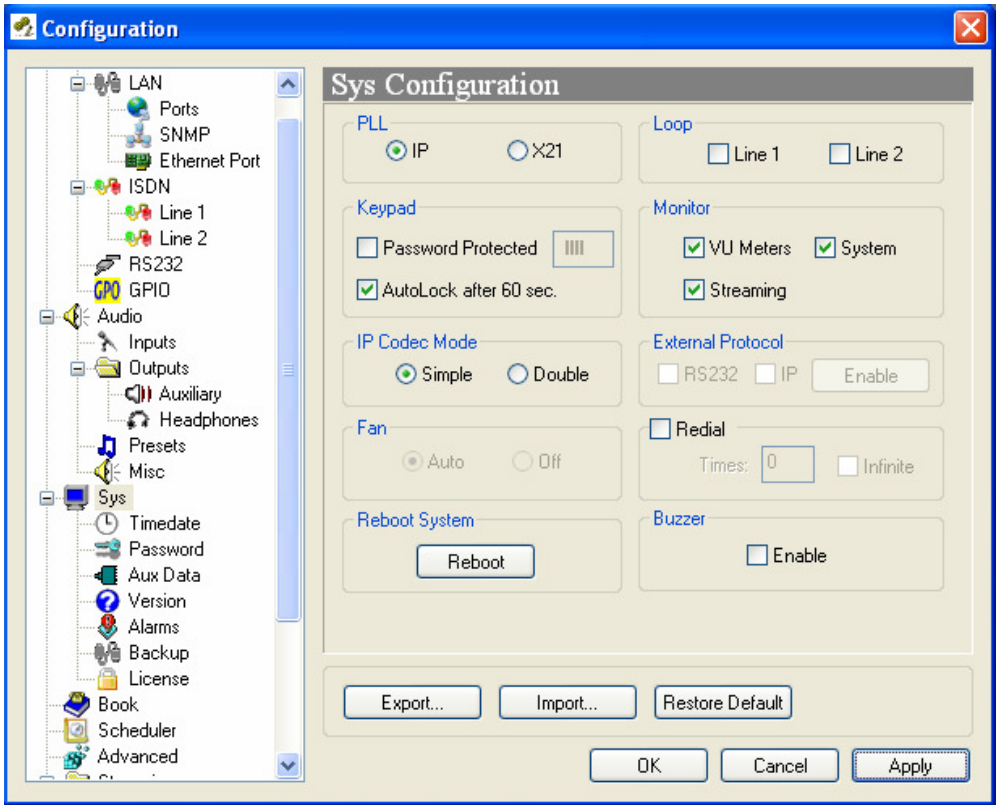


- **Monitor**: This option allows the user to hide or display additional information on the main window:

  - VU meters: Indicates audio levels.
  - System: Monitor voltages, temperatures, and fan operation.
  - Streaming: Shows clock-sync algorithm operation and Buffer Occupation Graph (Real Time Network Analyzer)[11]. This algorithm comes into scene when receiving audio over IP. This display is active when the audio data is "framed" - Under

[11] For more information please read chapter IX.6.7 – Prodys Proprietary Protocol v2.

normal circumstances the indicator should be in the middle Green area.



- **Reboot System**: A new option to reset remotely a unit was included in the SYS configuration menu.

- **PortaNet external protocol**: The PortaNet Software Development Kit provides a tool to manage the PortaNet units from external applications or devices different than ProdysControl. In this manner, it is possible to customize the control of the units and to integrate them in a global management system.

  The protocol is based on which can be managed through two different communication ports: the LAN port and the RS232 B port. For more information, please read ProntoNet SDK User's Manual.

- **IP Codec Mode**: There are two different operation modes: SIMPLE and DOUBLE mode. In SIMPLE mode, the unit will cannot establish two independent communications via IP. In DOUBLE mode, the unit can establish two independent communications via IP.

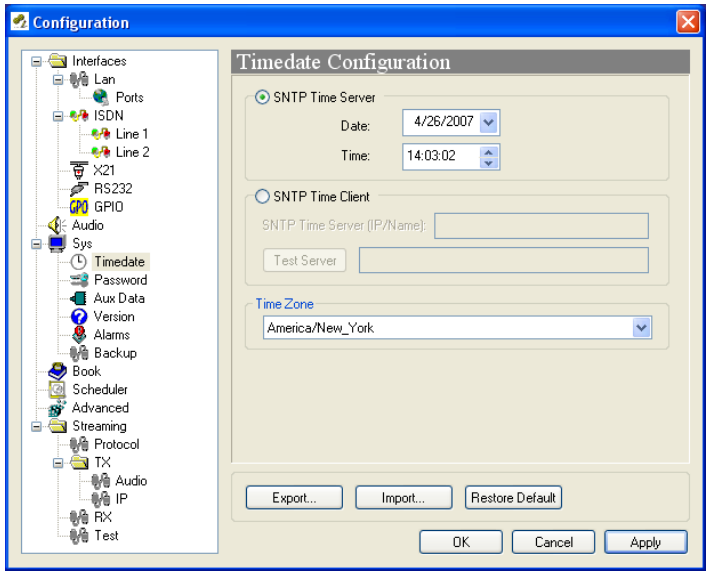
  In DOUBLE mode over IP, the following restrictions will be applied:

1.- Encoder 1 and Encoder 2 can only be set in MONO.
2.- Encoder 2 (assigned to line 2) can only be set in G722.

- **Buzzer**: It is possible to enable/disable the buzzer from this option. The buffer warns the user about incoming calls when the unit is configured as an ISDN codec and the answer mode is set to manual.

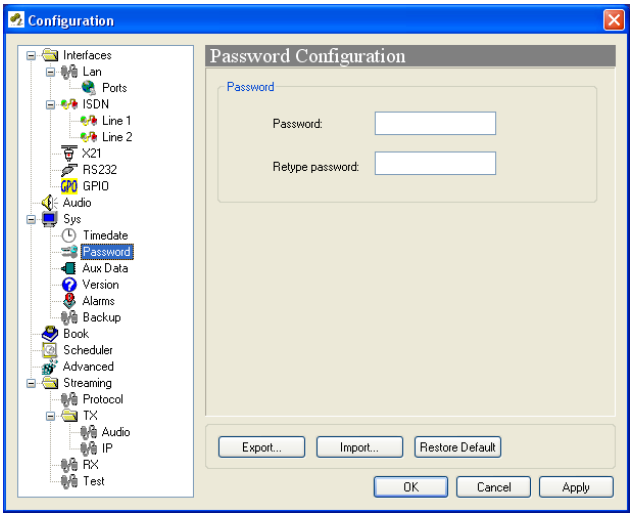
VIII.2.2.1 TimeDate

This option allows the user to synchronize time and date by using SNTP protocol (Simple Network Time Protocol). SNTP operates always in the client-server model and for this reason, PortaNet can work as SNTP server or SNTP client.



VIII.2.2.2 Password

Whenever the web browser is started, it is necessary to enter a security password. This is the dialog to program it.



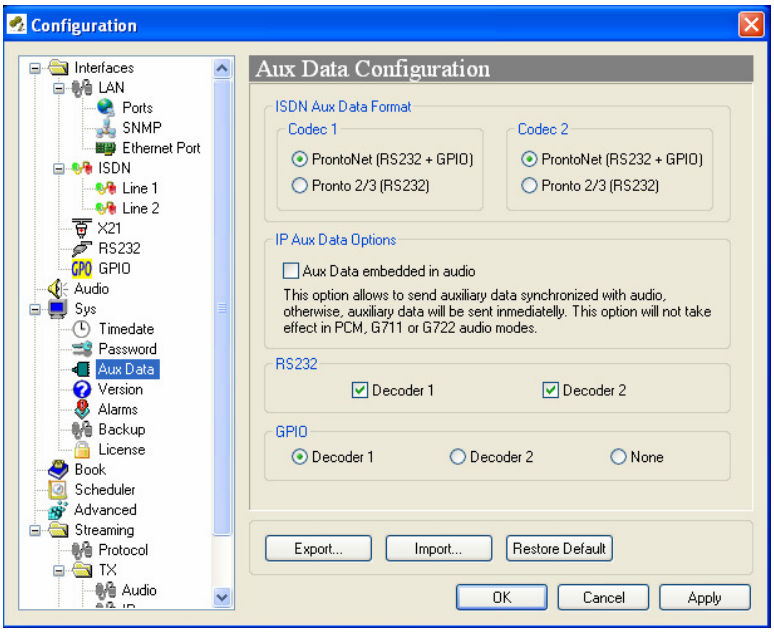
VIII.2.2.3 Aux Data

- RS232: There is one RS232 port for use as auxiliary data port. This port allows the transmission and reception of data along with the encoded audio. When the option is checked, the corresponding decoder will allow the reception of ancillary data.

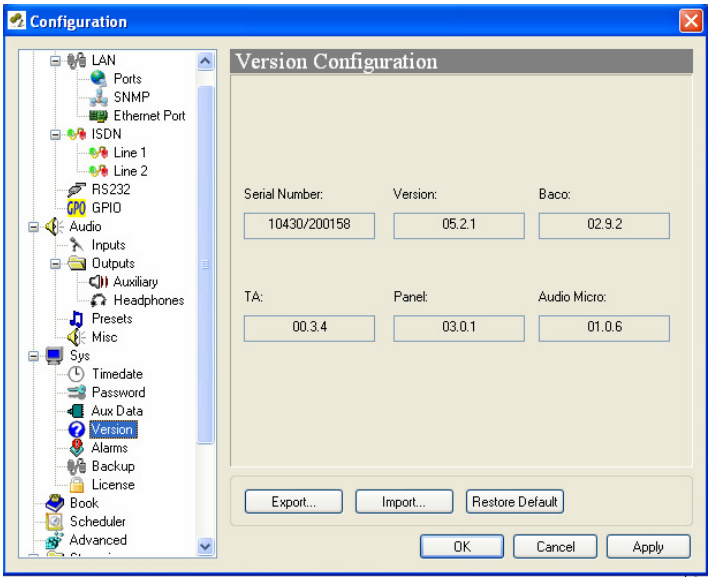
  Over ISDN, it is possible to choose between ProntoNet format and Pronto2/3 format (in which the GPIOs information cannot be sent). To be backward compatible with ISDN Pronto2/3 Prodys devices, select Pronto2/3 format.

  When selecting IP as communication interface, the auxiliary data can be sent via a different path, different from the audio one (the auxiliary data are embedded in the audio frames). This method has 3 big advantages: less delay independent from the audio codification delay; the possibility to send/receive auxiliary data regardless of the compression mode used for the audio communication; and the possibility to send/receive the User bit of the AES/EBU frame. The drawback is that the audio and data delay won't be the same.

- GPO: Under 'transparent' configuration, the state of the input will be present in its homologous output in the device connected at the other end. Since there is only one GPO connector, user must select whether to select GPO information coming from Decoder1 or from Decoder2, or none.

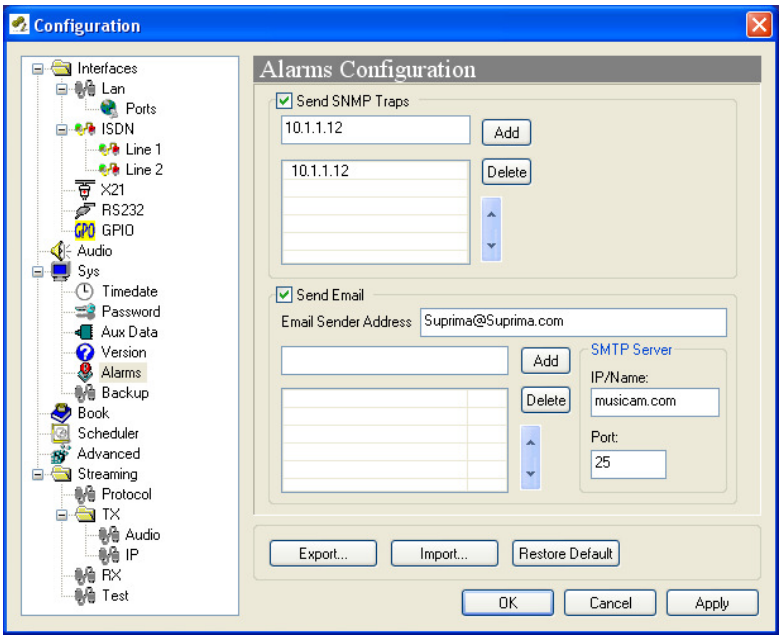


VIII.2.2.4 Software Versions



VIII.2.2.5 Alarms

From this option it is possible to select the IP address where the SNMP traps will be sent. An SNMP manager located at this IP address will receive and process the information according to the SNMP protocol. These SNMP traps will notify alarm information, that is, an SNMP trap will be sent when an alarm is activated or deactivated, with the time and type of alarm. For this to happen, alarms have to be enabled (see chapter VIII.8 Alarms). PortaNet can be fully monitored though SNMP protocol. It is also possible to notify alarm events via email. The user can send alarm information to several email addresses at the same time.



VIII.2.2.6 Backup

Backup Configuration. The PortaNet allows the user to use the ISDN line as a backup when the unit works in IP mode.

The backup is enabled by clicking the "Backup" box.

> **For more information about the backup operation see chapter IX.10 – How the backup Works?**

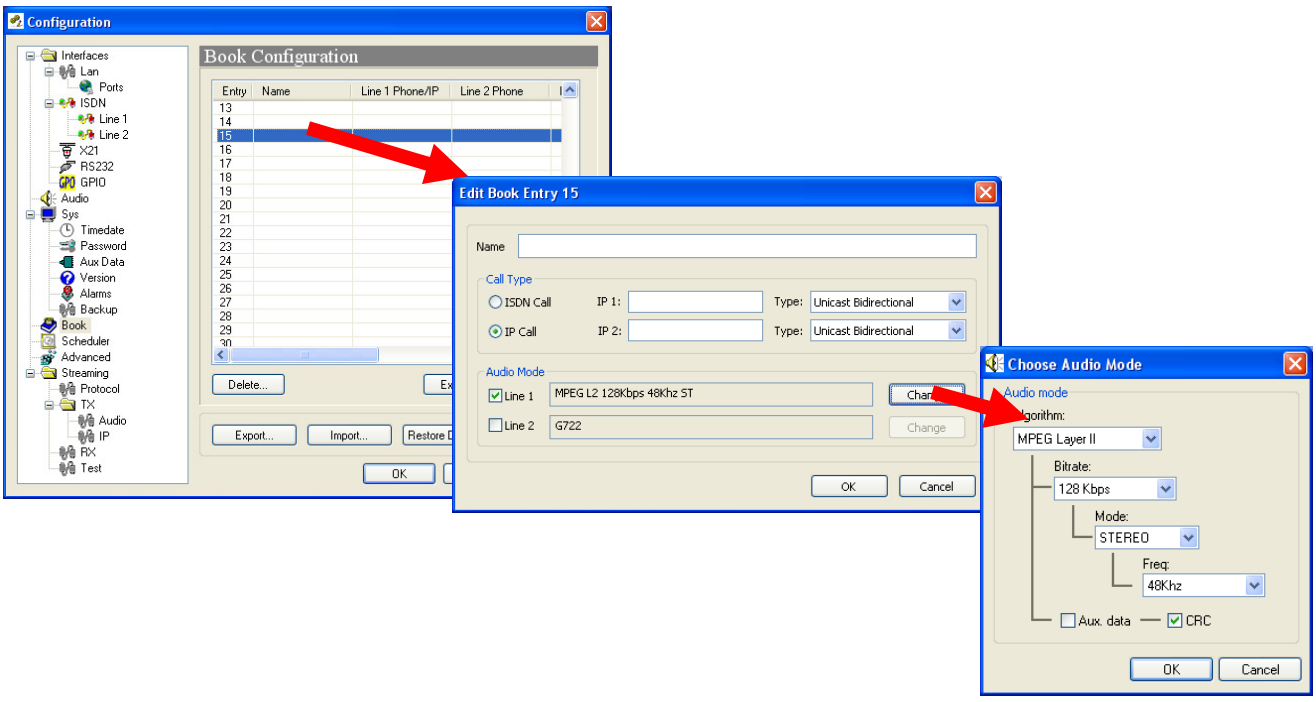
VIII.2.2.7 Phone Book

When the BOOK option is selected, the phone book window is showed on the right side. The Phone Book records the user name, a telephone number (optionally two), an IP address. There are up to 32 indexes.

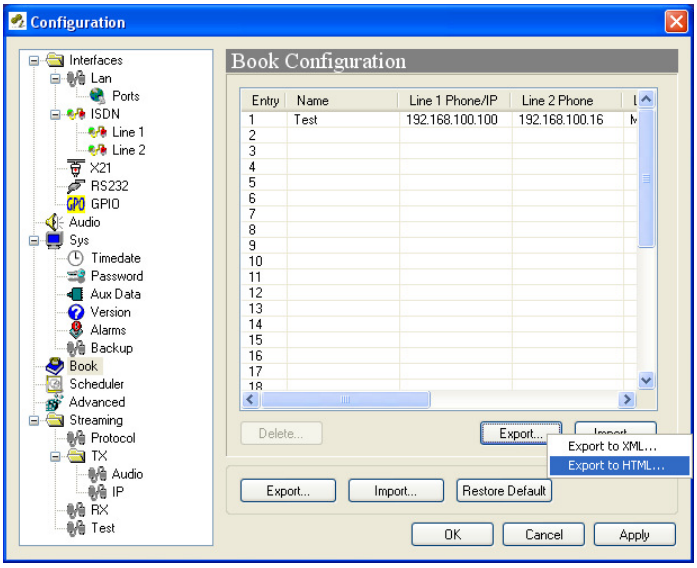
> **It is not possible to store ISDN numbers and IP address in the same index.**

The proceeding to edit a phone book index is as follows:

1. To select one index by clicking on its area.
2. To select ISDN numbers or IP Address depending on which network will be used. Only it is possible to select one of them.
3. To enter the ISDN number/numbers or the IP address according to the before selection.
4. To configure the encoder is optional. In case of that it is not entered, the unit will proceed to call in the current encoder configuration.



It is possible to Export / Import the Book in XML format. It is also possible to create an html report allowing to preview and print.



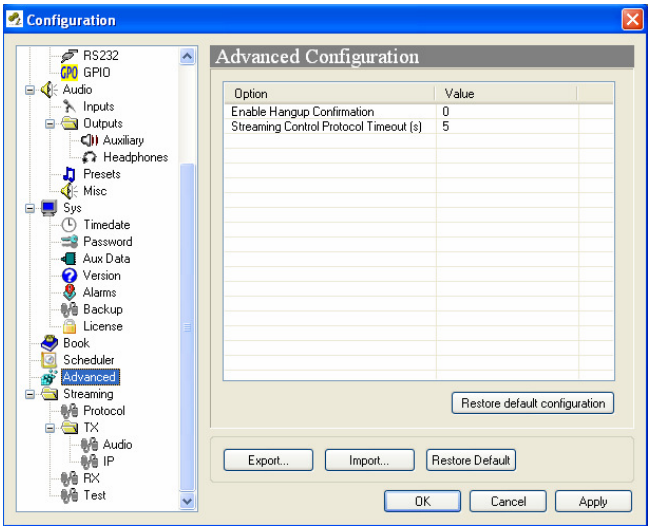
The audio compression mode is restored to that one prior to the connection. This allows the user to change the compression mode during the connection, or to make calls through the book entries with compression modes different from the former one.

VIII.2.2.8 Scheduler

The Scheduler allows the user to program calls to be made automatically. For more information read chapter VIII.6 - Scheduler.

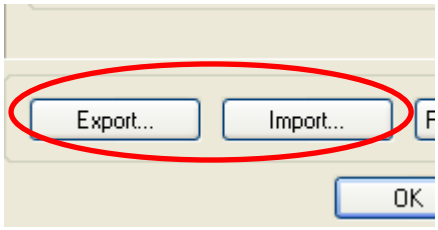
VIII.2.2.9 Advanced

Streaming Multicast Rejoin On Timeout: IGMP protocol functionality.
Enable Hangup Confirmation: If the value is "1", the application will ask confirmation before disconeccting the line.
Streaming Control Protocol Timeout: When the link is down for the specified number of seconds, the audio connection is finished automatically by the unit. This avoid situations where the connection is lost, and the unit still shows 'connected' on the screen.



VIII.2.2.10 Exporting / Importing the configuration

It is possible to export and import the whole system configuration:



VIII.2.3 Streaming

Streaming parameters objectives:

-Provide a 'test' tool to check in real time the bandwidth, delay and jitter of the IP connection between two PortaNets.

-To allow adjustment of the various transmission and reception parameters to optimize the unit to provide the best quality real-time audio streaming for a particular network connection.

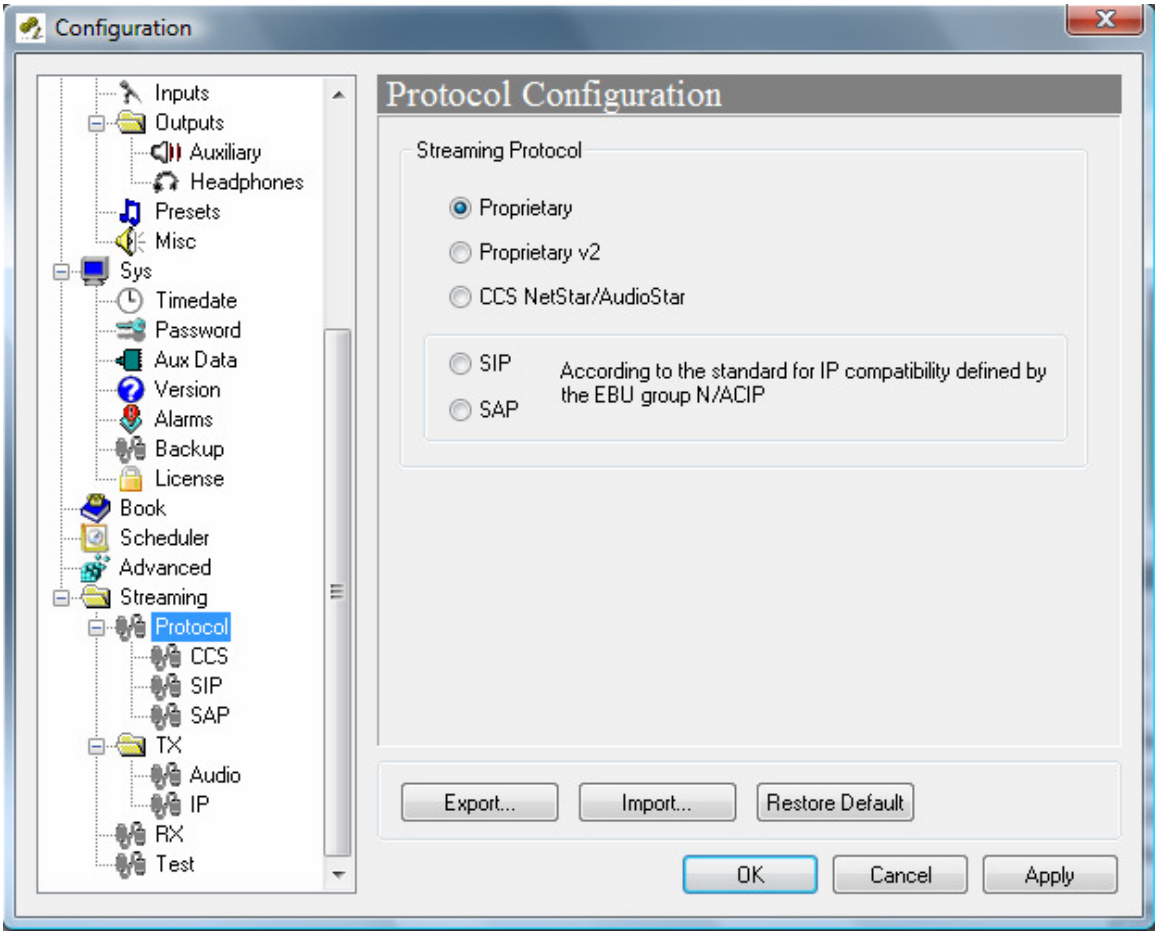
These tools are grouped in the following manner:

1.- Protocol.
2.- Transmission parameters.
3.- Reception parameters.
4.- Test tool.

VIII.2.3.1 Protocol

Streaming Protocol: This option allows the user to configure the streaming protocol to use for transmission over IP. There are five choices;

- Proprietary: Prodys proprietary IP streaming protocol.
- Proprietary v2: Prodys proprietary IP streaming protocol version 2.
- SIP: SIP is a signaling protocol for creating, modifying, and terminating sessions with one or more participants. For more information please check the proper section of this document.
- SAP: SAP is a protocol for broadcasting multicast session information.
- CCS NetStart/AudioStar.



VIII.2.3.2 Tx

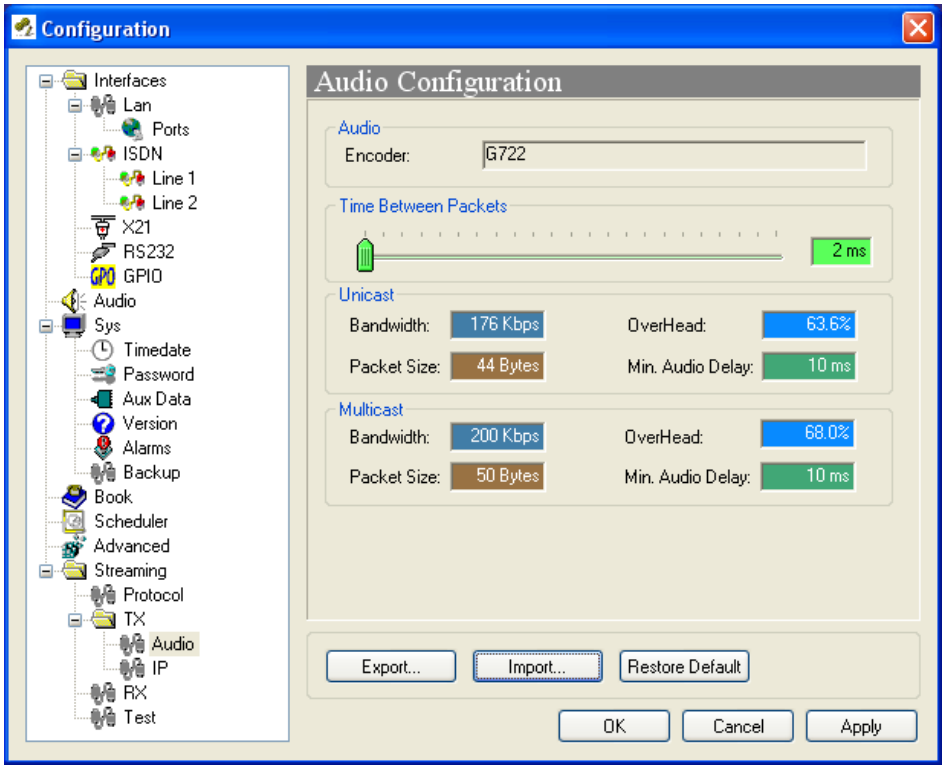
Audio: This option allows the user to configure the audio transmission parameter (TBP: Time between packets) and to know in advance, the required bandwidth and delay of the connection.

IP: This option allows the user to configure some IP header fields to carry out QoS.

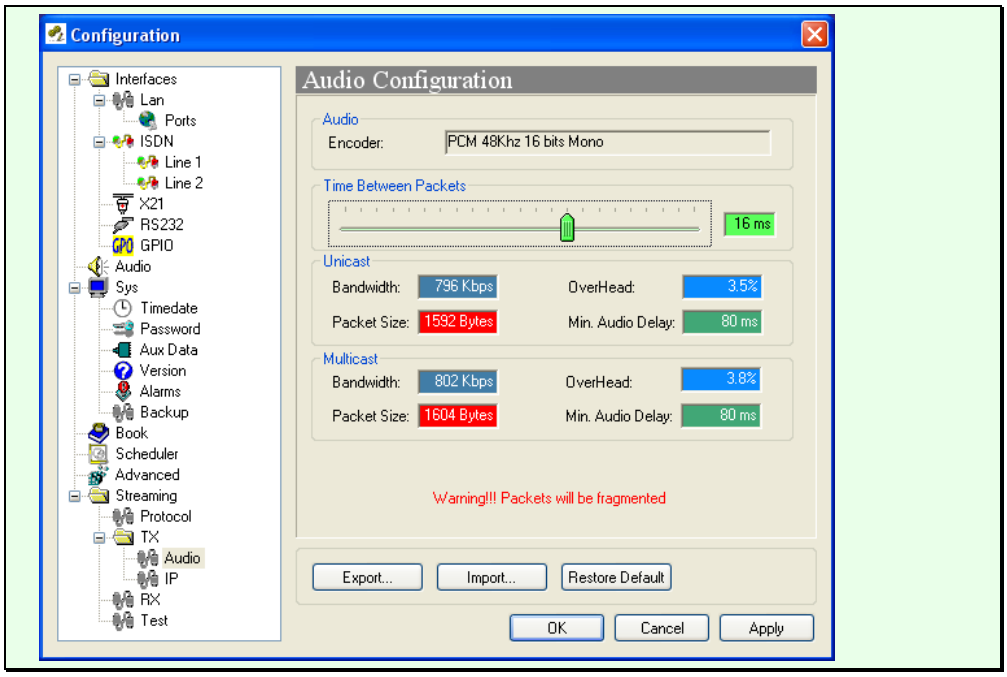
VIII.2.3.2.1 Audio

The 'Time Between Packets' (TBP) parameter is directly related with the packet size and therefore with the occupied bandwidth (overhead) and delay. Therefore, the appropriate value for this parameter is a trade off between delay and bandwidth.

In order to achieve the minimum delay between encoder and decoder it is necessary that the transmission time between packets is minimum as well. However, the greater the TBP, the less bandwidth is required (less overhead), but the more delay there will be in the connection. Otherwise, the smaller the TBP, the smaller the packet size, and the greater the overhead, due to the IP headers in the packets. With this tool, the user can know beforehand the required bandwidth, overhead, packet size and delay for the current compression mode and TBP value.



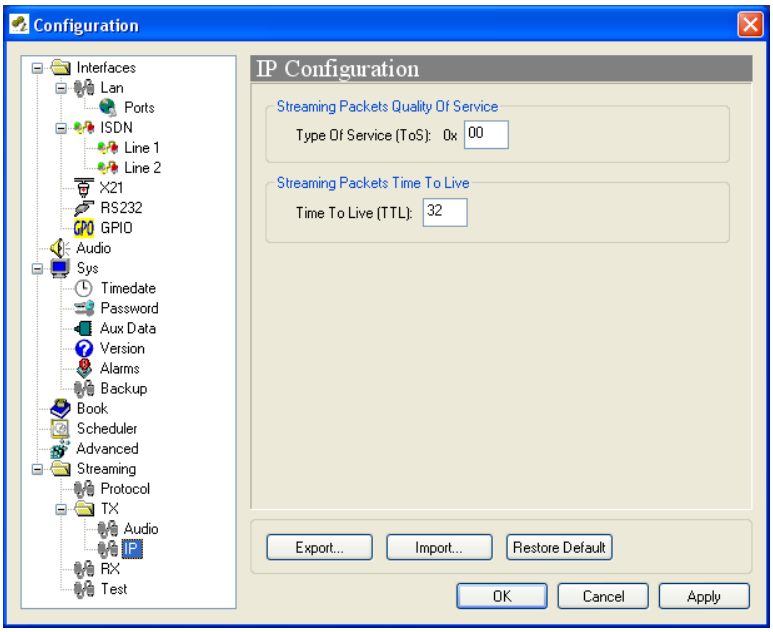
When the packet size needs to be fragmented, keep in mind that the overhead changes for that reason. When it happens it is indicated as follows:



VIII.2.3.2.2 IP:

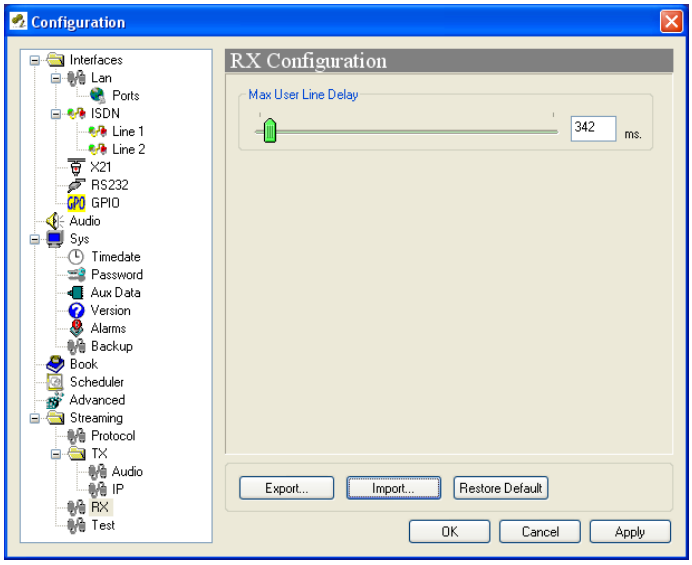
This option allows the user to configure some IP header fields to carry out QoS.

- Streaming Packets Quality of Service: This option allows the user to configure the value in the ToS field in the IP header. This field can be used to carry out QoS in the network.
- Streaming packets Time To Live: Configure the Time To Live parameter.



VIII.2.3.3 Rx

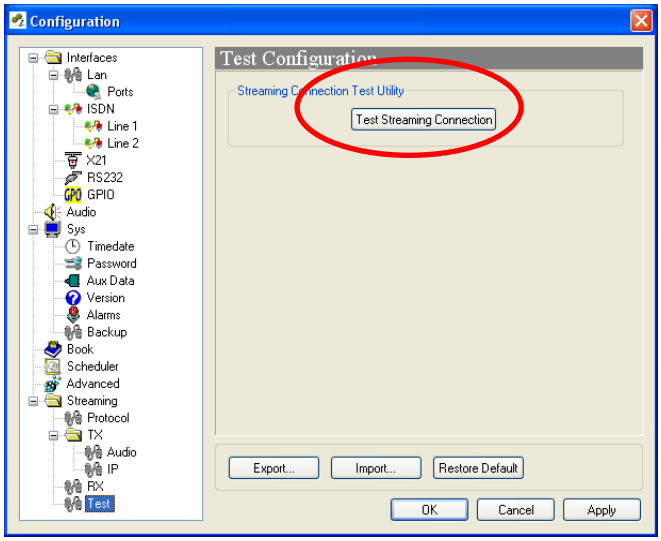
A decisive factor in real time audio streaming is the 'jitter', or delay variation. To deal with the jitter in the connection, PortaNet provides a tool which allows the user to modify the size of the reception buffer, and so, to compensate the jitter. The maximum value for this buffer is 10 sc.

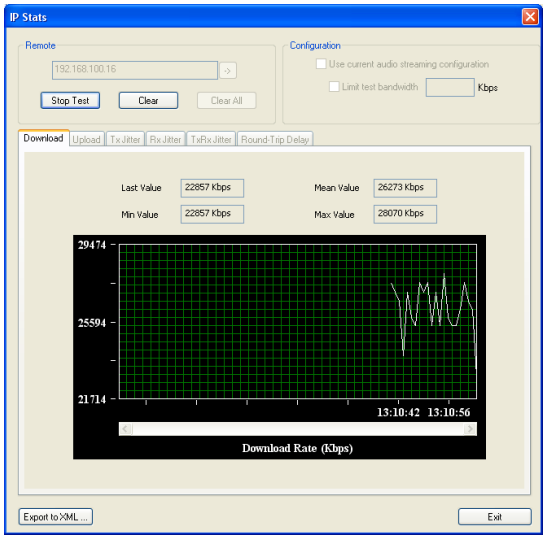


This value can be changed during the audio connection.

VIII.2.3.4 Test

This tool allows the user to check, in real time the upload/download bandwidth, delay, jitter and packet loss in an IP link between two PortaNets. This information will be used to adjust the streaming parameters in order to achieve the best quality in the audio streaming connection.

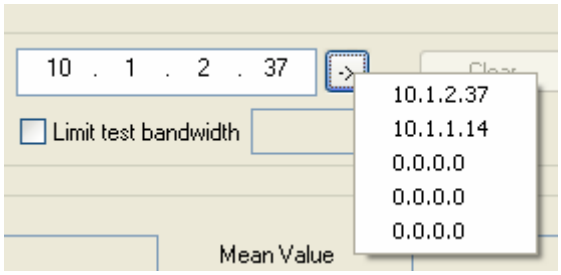




This tool is only available when lines are disconnected.

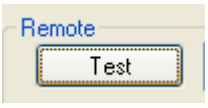
How to run the tests**:**

1.- First of all the user must enter the IP address of the remote PortaNet. This dialog stores the last used IP addresses.



2.- Select the test you wish to perform.

3.- To run the test just press the test button:

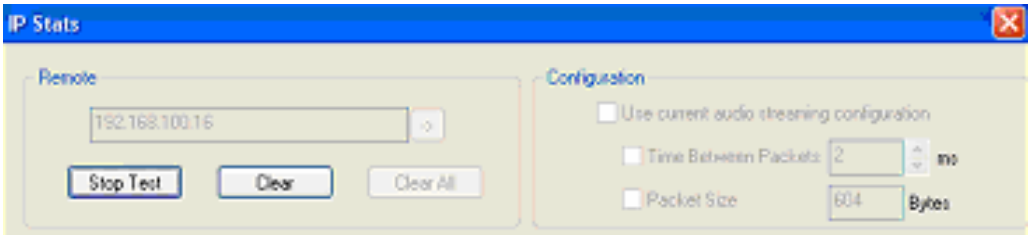


4.- The values will be shown in a graph, as well as the max, min, last, and mean(avg) values.

4.- Press "STOP" to stop the test or "CLEAR" to reset the readings.

5.- The user can limit the bandwidth used by the test tool:

By default this test will attempt to send as much IP traffic as possible. If you are using this test on a "live" network then it is desirable to limit the test bandwidth to that which will actually be used. It is also possible to simulate the same packet rate and bandwidth as that configured for the audio connection[12].

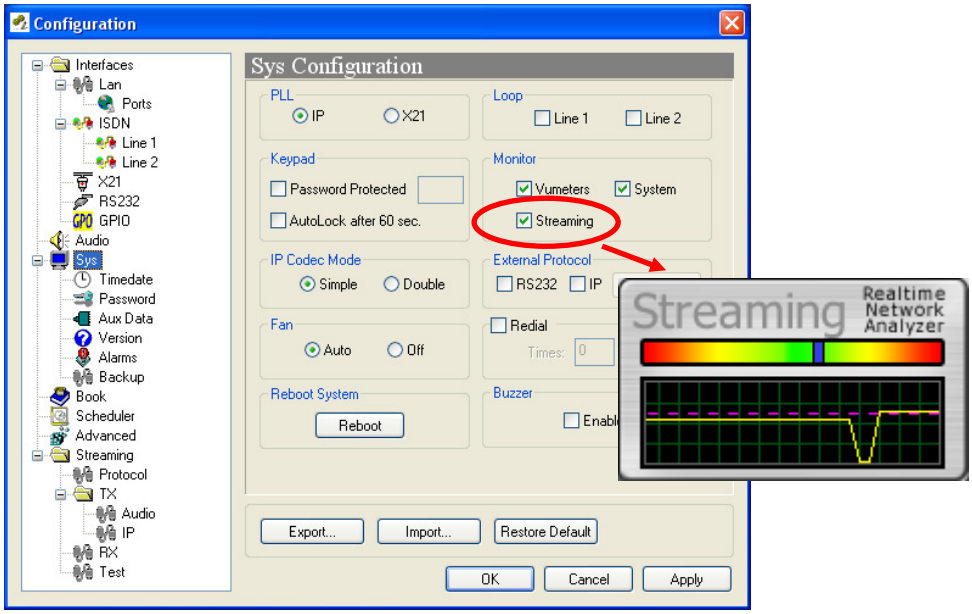


An accurate figure of total bandwidth used (audio data + overhead) can be obtained from the "Audio Configuaration" screen.

Once the test is finished, it is possible to export the results to an XML file.

VIII.2.3.5 Real Time Monitoring

It is possible to monitor the streaming operation in real time by ticking a checkbox.



[12] Available from version 5.0.0 onwards.

That indication represents the instant decoder buffer occupation.

When indication is green or even yellow means the streaming is working fine. Depending on the algorithm, the variation will be quicker or slower but always should be going up and down from yellow to the center (green).

If the bar moves towards the red area, it means that it is getting closer to buffer overflow (red right) or buffer underflow (red left). Of course, if you have buffer overflow or underflow you will have a cut in the audio. If the indication goes to red for an instant, usually doesn't affect the streaming, but of course, if you have a constant red indication you are having problems with the streaming.

Buffer underflow might be caused by lost packets or because the clock of the AD converter at the transmitter side which runs slower than the DA converter at the receiver side.

Buffer overflow might be caused by the opposite scheme: The clock at the transmitter side is running quicker than the clock at the receiver side.

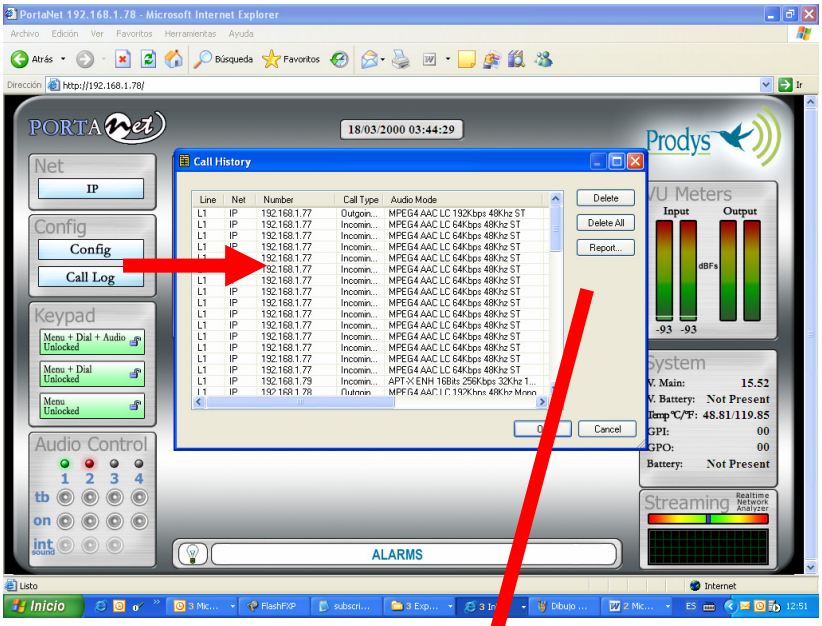
Prodys PPL adjustment mechanism is aimed at dealing with this clock jitter and thus to compensate for this variation and guarantee proper audio over IP.

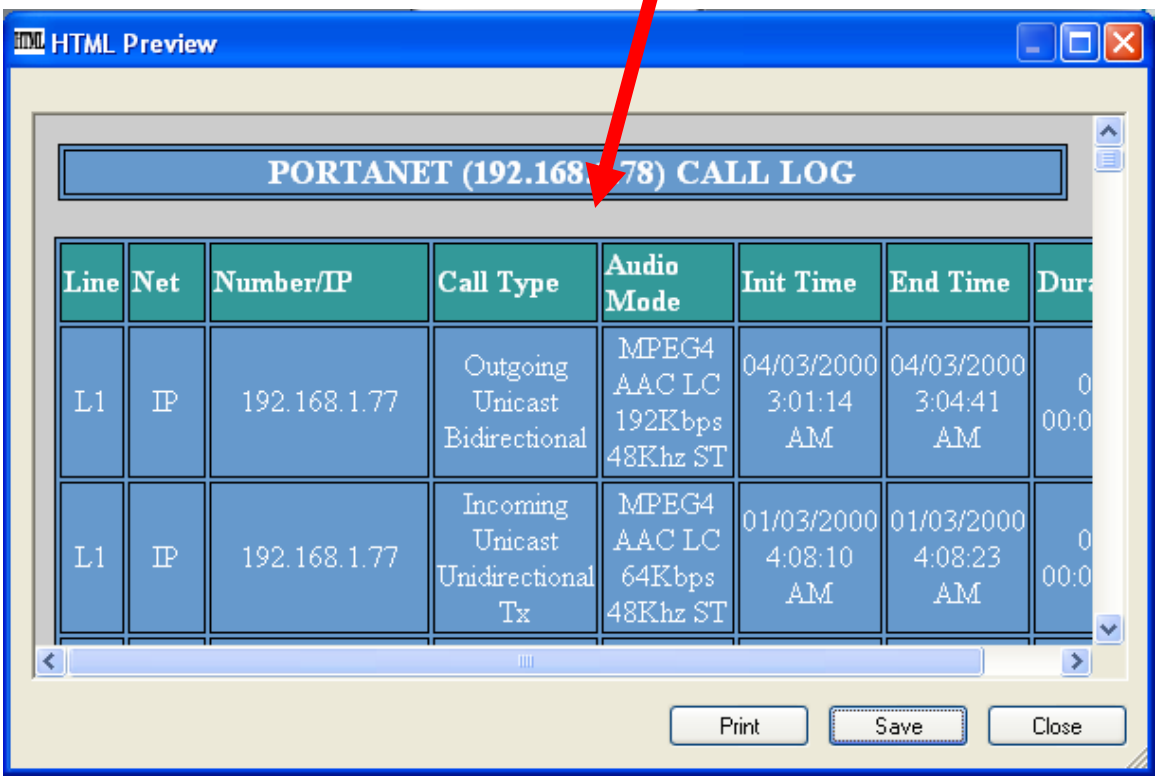
This option also enables the Real Time Analyzer control graph under the streaming bar. By clicking on the graph, the real time analyzer window will appear. For more information about the real time analyzer please read chapter IX.6.7 - Proprietary set of Protocols v2.

VIII.3 Call Log

An history report has been included to record the input and output calls according to the following information: telephone/IP, audio modes, date and start/end time, length of each call, etc…

A report of the calls in html can be created allowing preview and print.


PortaNet 192.168.1.78 - Microsoft Internet Explorer
PORTANET
18/03/2000 03:44:29
Prodys
Net
IP
Config
Config
Call Log
Keypad
Menu + Dial + Audio Unlocked
Menu + Dial Unlocked
Menu Unlocked
Audio Control
Call History
Line
Net
Number
Call Type
Audio Mode
Delete
Delete All
Report...
Cancel
VU Meters
Input
Output
dBFs
-93 -93
System
V. Main: 15.52
V. Battery: Not Present
Temp °C/°F: 48.81/119.85
GPI: 00
GPO: 00
Battery: Not Present
Streaming
ALARMS

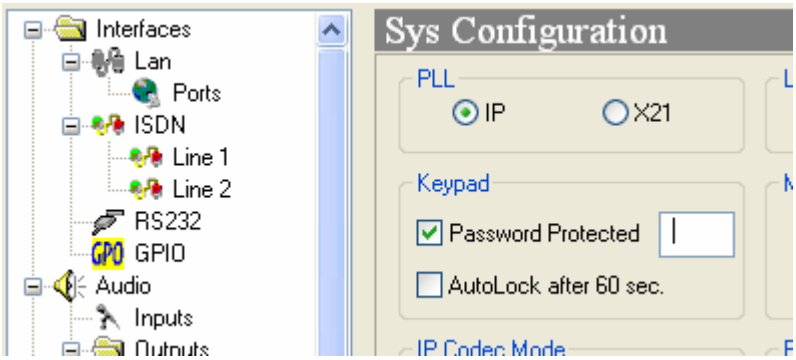

HTML Preview
PORTANET (192.168.1.78) CALL LOG
Line
Net
Number/IP
Call Type
Audio Mode
Init Time
End Time
L1
IP
192.168.1.77
Outgoing Unicast Bidirectional
MPEG4 AAC LC 192Kbps 48Khz ST
04/03/2000 3:01:14 AM
04/03/2000 3:04:41 AM
L1
IP
192.168.1.77
Incoming Unicast Unidirectional Tx
MPEG4 AAC LC 64Kbps 48Khz ST
01/03/2000 4:08:10 AM
01/03/2000 4:08:23 AM
Print
Save
Close

VIII.4 Keypad locking

It is possible to lock the PortaNet keypad remotely. This can be done in three different ways:

a. **Menu+ Dial + Audio** → To lock all the keys on the top panel.
b. **Menu + Dial** → To lock only the dial and menu keypad, but not the audio control keypad.
c. **Menu** → To Lock only menu keypad, but not the dial and audio control keypad.

Moreover, there are two more options related with the keypad locking in the Sys configuration menu:



- **Password protected**: This option allows to set up a password that will be necessary to unlock the keypad from the PortaNet keyboard.
- **AutoLock after 60 sec**: Once this option is enabled, the complete keyboard will be locked after 60 seconds without activity.

These options are also availble in the display menu:

SYS → KEYPAD → AUTOLOCK: ENABLE /DISABLE.

SYS → KEYPAD → PASSWORD: ENABLE + PASSWORD

VIII.4.1 Locking the keypad from the Portanet keyboard

To Lock the keypad from the PortaNet top panel keyboard, use the following key combinations:

- **"OK" + "*"** → To lock all the keys on the top surface.
- **"OK" + "#"** → To lock only the dial and menu keypad, but not the audio control keypad.
- **"OK" + "0"** → To lock only menu keypad, but not the dial and audio control keypad.

To unlock just press consecutively **"OK" + "*"**.

VIII.5 Audio control keypad

The audio control keypad can be controlled/monitored remotely from the web page.

ON control

Each MIC/LINE audio input has a key to enable/disable it. When an input is enabled, it will contribute to the audio being sent through the program line (line 1). The corresponding LED will light on Green, indicating that the input is enabled and going through the line 1.

TB control

PortaNet supports two independent communications via IP or ISDN. Thus, we can establish one connection for the main program, and another one for coordination, or TalkBack program. TB keys allow the connection of the corresponding MIC/LINE audio input to the talkback channel. The corresponding LED will light on RED.

Int. Sound control

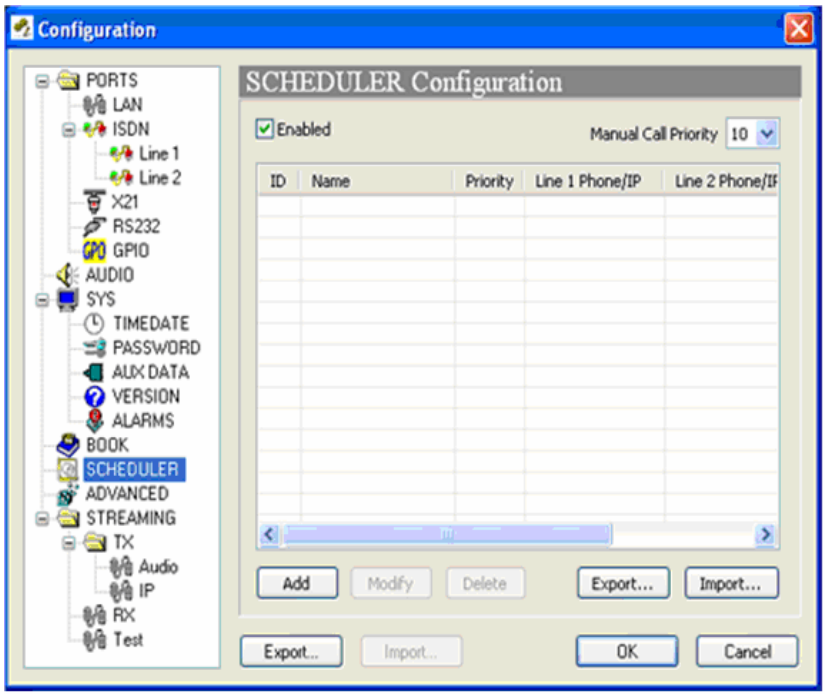
There is an additional LINE level audio input that can be used to feed the headphone outputs. Each headphone output has a control to enable or disable this signal.

VIII.6 Scheduler

The Scheduler allows the user to program calls to be made automatically. In addition, the duration and the frequency for each call can be defined.

VIII.6.1 Configuration

To access the Scheduler configuration window, the user has to click on 'Config' on the web page, and then select "Scheduler" in the general configuration window.



VIII.6.2 How to enable/disable the Scheduler

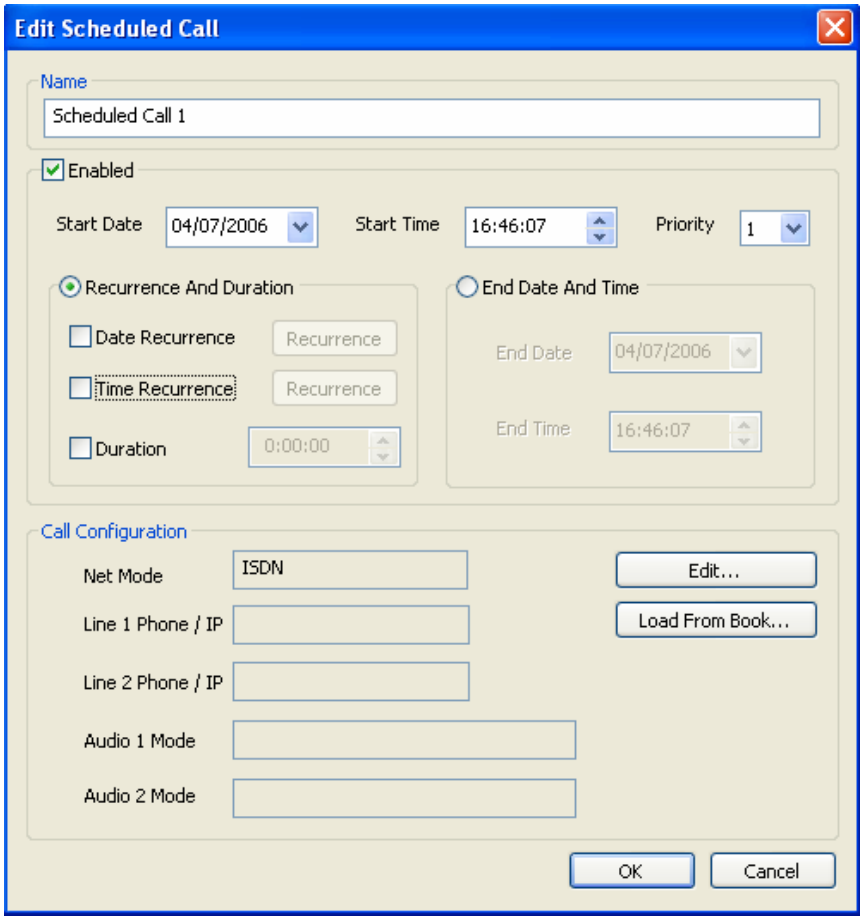
The Scheduler dialog has a checkbox which allows the activation/deactivation of the Scheduler. When the Scheduler system is disabled, none of the scheduled calls will be made. In addition, each scheduled call can be enabled/disabled individually.

VIII.6.3 Automatic and manual call

It is possible to define the precedence between manual and automatic calls. Thus, the user can decide if one automatic call will take precedence over a manual one. The call priority can be set to a number between 1 and 10 (highest priority). The Scheduler will only establish a programmed call when this call has the same or higher priority than the current one. The default priority for automatic calls is 1, and the default one for manual calls is 10. Manuals calls are those made by the user. It is possible to modify the default priority for the manual calls from the Scheduler dialog, by changing the value of the field 'Manual call priority'.

VIII.6.4 Programming the scheduler

To add a new automatic call, click on the 'Add' button in the Scheduler configuration window. The following window will be displayed:



This dialog permits the user to configure all the parameters related to an automatic call.

VIII.6.5 Name

This field allows the user to enter a brief description to identify this call. This name will appear in the call list of the Scheduler. By default, the name will be "Scheduled Call" followed by a number.

VIII.6.6 Start

The "Start Date" and "Start Time" fields permit the user to configure the date and time when the scheduled call or pattern will be enabled.

VIII.6.7 Priority

By clicking on "Priority" the user can change the priority value assigned to the current entry. More information about priorities can be obtained in chapter 2.2.

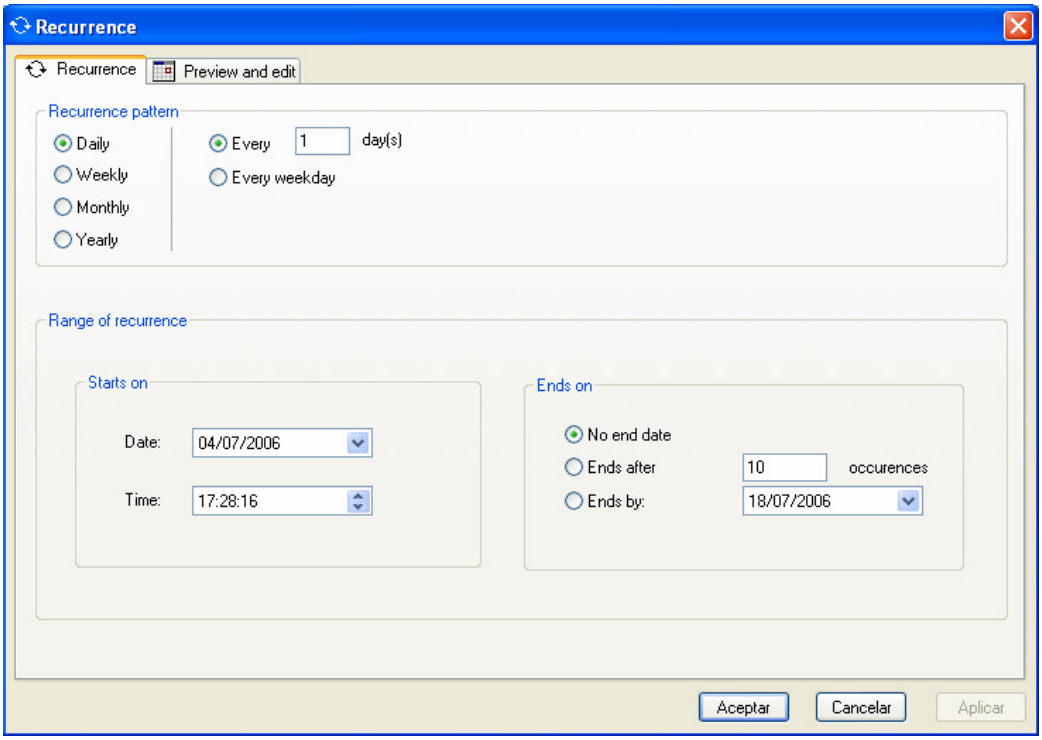
VIII.6.8 End

There are two methods to define how long the call will last: "Recurrence and duration" and "End date and Time".

- **"Recurrence and duration"**

This option allows the programming of call repetitions based on time or date: Time recurrence and Date recurrence.

To configure the unit for "Date recurrence" mark the corresponding checkbox in the Scheduler configuration window and click on "Recurrence". The following window will be displayed:



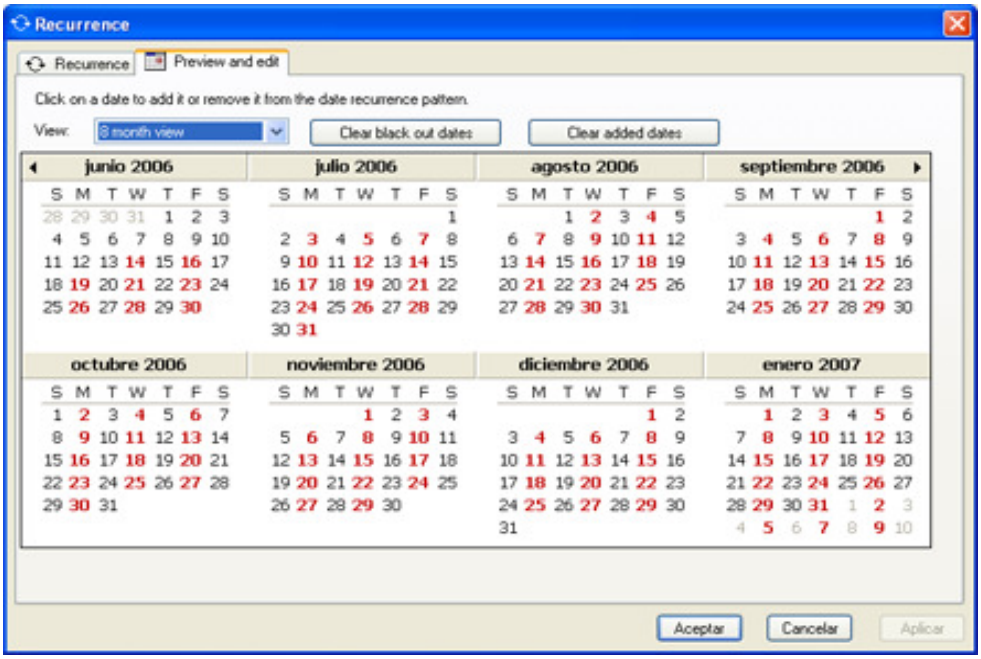
This window is arranged in two different sections:

- "Recurrence pattern": This part allows the user to configure the dates when the call
  will be made. It is possible to program daily, weekly, monthly and yearly patterns.

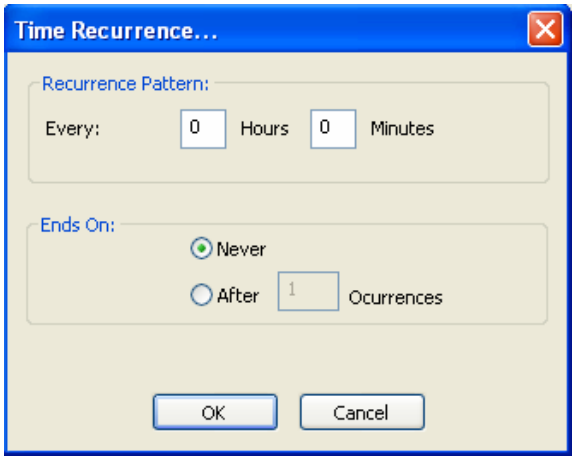
- "Range of recurrence": This section is divided in two different parts:

  - "Starts on": This option establish the start date and time for the configured pattern.
  - "Ends on": It is possible to establish the end of the configured pattern in three different ways: Without end date, after a certain number of repetitions, or in an specific date.

By clicking on "Preview and edit" tab, the user will obtain a calendar wherein those days when this pattern is active will be coloured on red.



To configure the unit for "Time recurrence" mark the corresponding checkbox in the Scheduler configuration window and click on "Recurrence". The following window will be displayed:



This window is arranged in two different sections:

- "Recurrence pattern": This section permits the user to configure the period of time for this call to be repeated.
- "Ends on": From here, it is possible to establish the end of this recurrence. It can be never or after an specific number of occurrences.

- **"End date and time"**

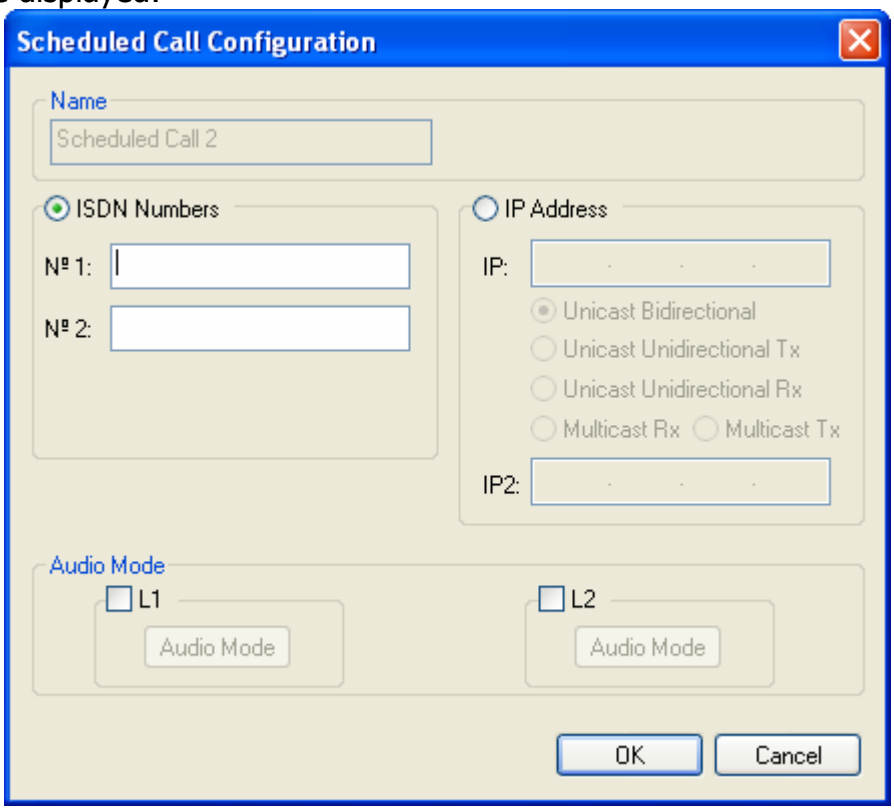
This option allows the user to establish the duration of a programmed call by defining the end time and date. The call will be established in the date and time defined by the **"**Start Date**"** and **"**Start Time**"** fields.

VIII.6.9 Scheduled call configuration

It is necessary to configure the type of network, the numbers and the compression mode that will be used for call establishment. To do that, the user can configure these parameters manually, or automatically, by getting them from one of the entries of the phone book.

VIII.6.9.1 Manual configuration

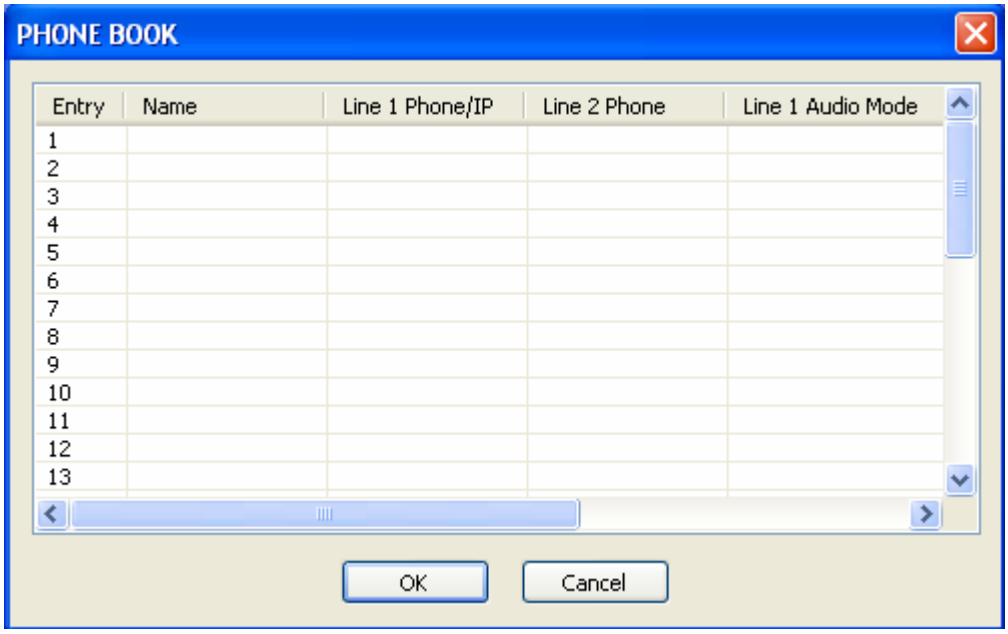
To configure these parameters manually, clic on "Edit…". The following dialog will be displayed:



From this window the user can enter all the parameters involved in a call establishment.

VIII.6.9.2 Automatic configuration from the phone book

To configure these parameters automatically, click on "Load From Book…" A window with all the entries of the phone book will appear. From this window the user can select any of the existing entries by clicking on any of them and on the OK button. The selected entry will be read and its configuration will be stored for the current pattern.



VIII.6.10 Modifying existing scheduled calls

To modify a pattern or scheduled call, the user has to select one scheduled call from the Scheduler list on the Scheduler Configuration Window. Then, by clicking on "Modify", the user will be able to edit all the parameters related to the select pattern.

VIII.6.11 Deleting existing scheduled calls

To delete a pattern or scheduled call, the user has to select one scheduled call from the Scheduler list on the Scheduler Configuration Window. Then, by clicking on "Delete", the select pattern will be deleted.

VIII.6.12 Copying and pasting scheduled calls

From the Scheduler configuration window, the user can copy and paste previously configured scheduled calls.

To copy a pattern, select one from the list and right-click on it and select the option "Copy Call(s)".

To paste a pattern, select one from the list and right-click on it and select the option "Paste Call(s)".

VIII.6.13 Monitoring scheduled calls

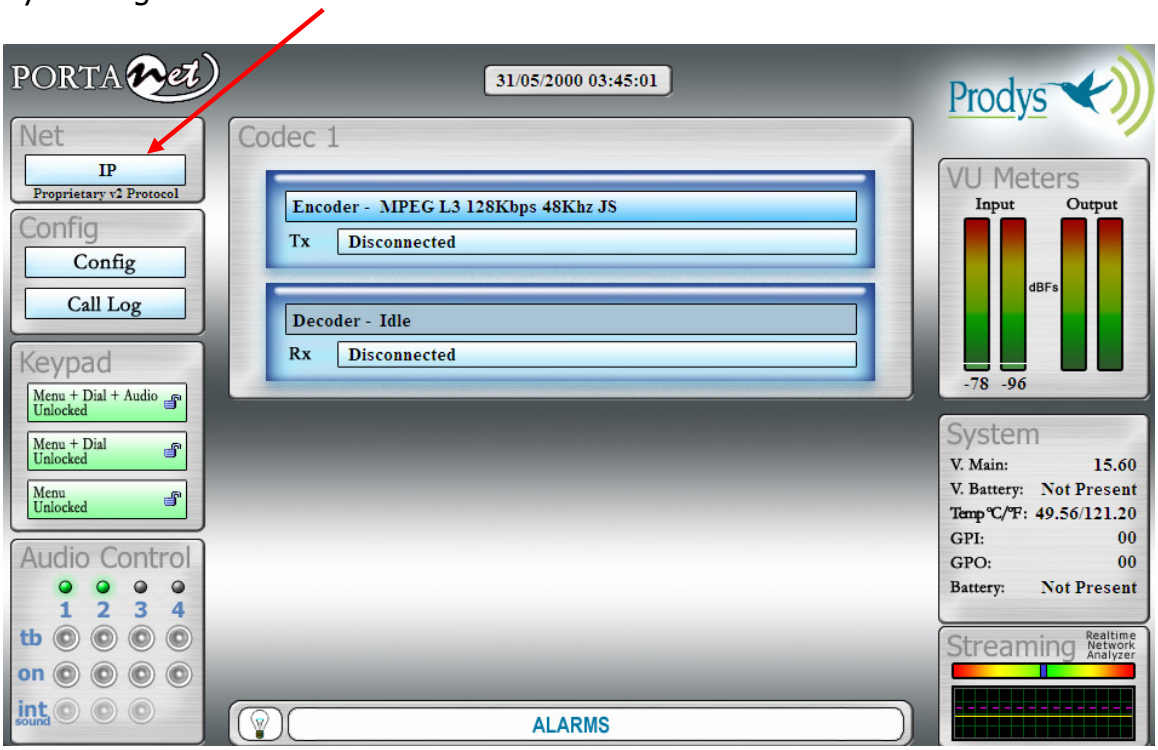
When one call related to one of the patterns defined in the scheduler of the unit is established, the message "SCHEDULED CALL IN PROGRESS" will be displayed on the web page for the duration of the call.

VIII.7 Controlling the PortaNet

From the control area it is possible to select the communication interface, as well as to configure the encoders and control the communications.

VIII.7.1 Selecting the NET interface

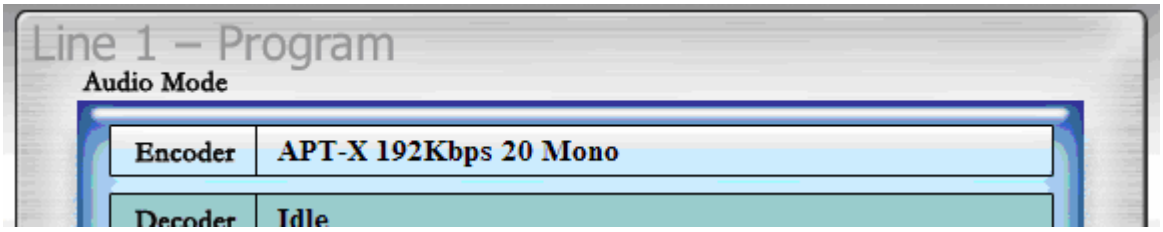
By clicking on the 'Net' area:



This option is only available when the lines are disconnected.

VIII.7.2 Configuring the Encoder

The 'Audio Mode' area shows the current encoder and decoder configuration. By double-clicking on the Encoder area the user can change the encoder configuration.

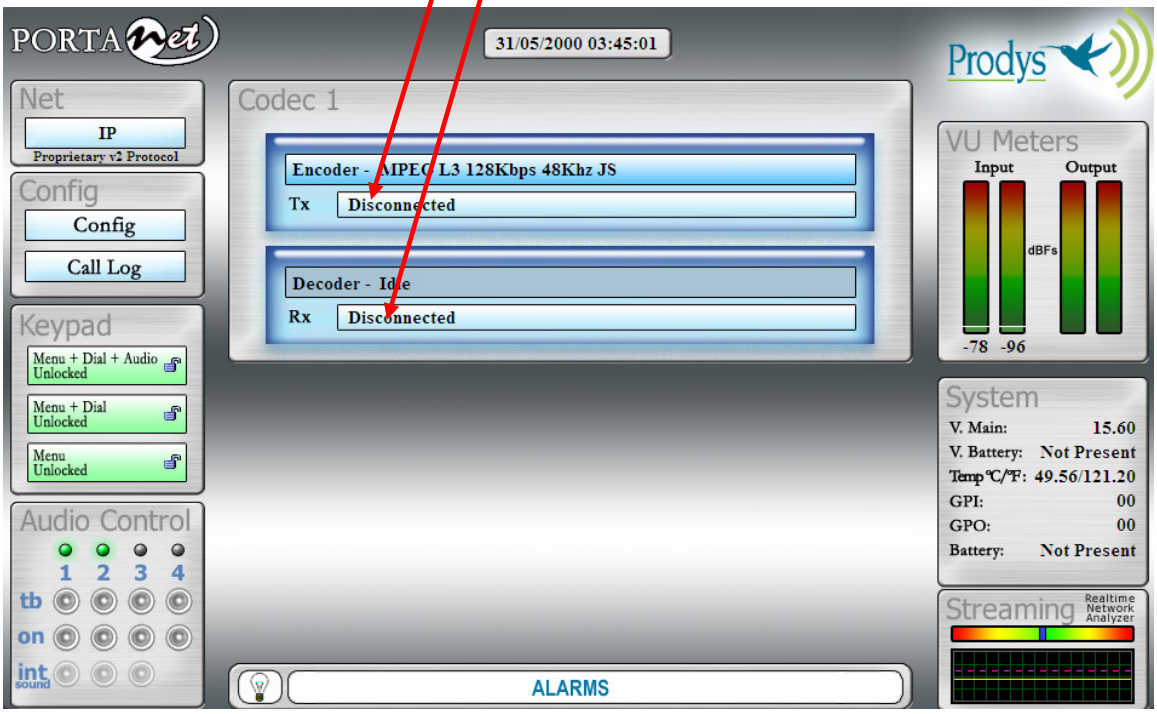


Over IP, the encoder of line 2 or TalkBack line is fixed in G722.

The audio compression mode is restored to that one prior to the connection. This allows the user to change the compression mode during the connection, or to make calls through the book entries with compression modes different from the former one.

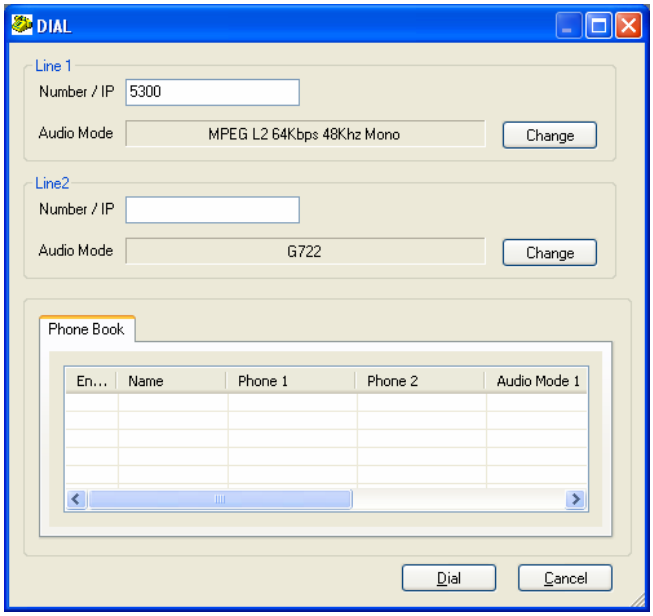
VIII.7.3 Making calls:

By double-clicking on the Tx or Rx connection area the user can access the call menu window:



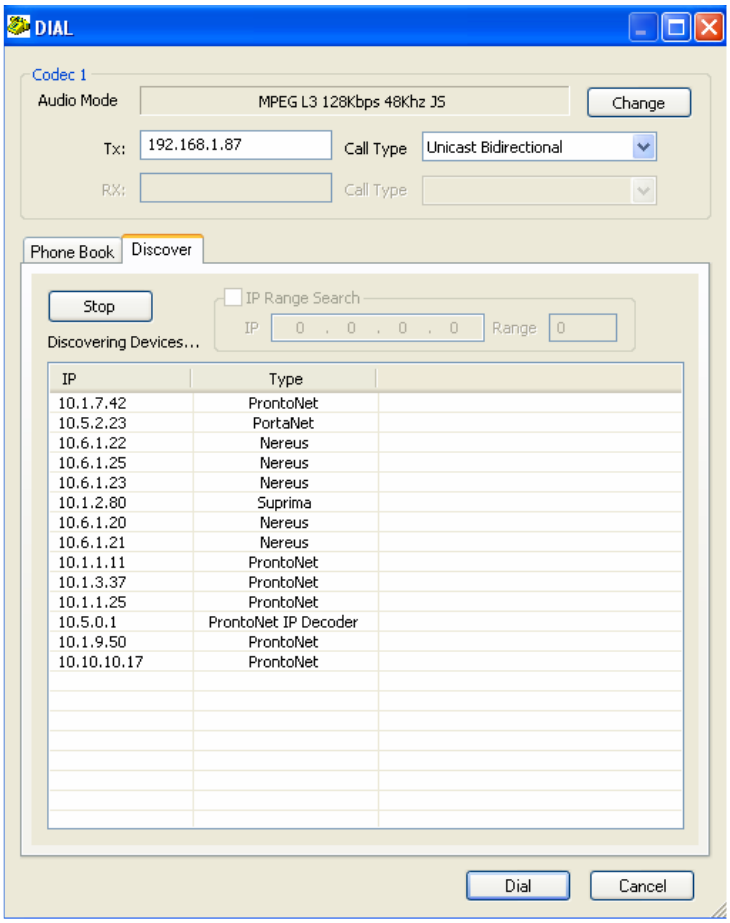
Depending on the selected communication interface, IP or ISDN, different options appear:

- When NET = ISDN, the following window is displayed:



The user can enter the ISDN number or can select an index from the Phone Book. The encoder can be configured from this dialog as well. To make the call click on "Dial".

- When NET = IP, the following window is displayed:



Call types:

- Unicast Bidirectional → This is a bidirectional point to point connection, that is, both ends will transmit and receive audio simultaneously. It will be necessary to check the upload and download bandwidth of the link.
- Unicast Unidirectional Tx → This is a unidirectional point to point communication where only the end which makes the call will send audio.
- Unicast Unidirectional Rx → This is a unidirectional point to point communication, where the calling end will be the receiver.
- Multicast Tx → This is point to multipoint communication in which the calling end will join a multicast group as a transmitter.
- Multicast Rx → This is point to multipoint communication in which the calling end will join a multicast group as a receiver.

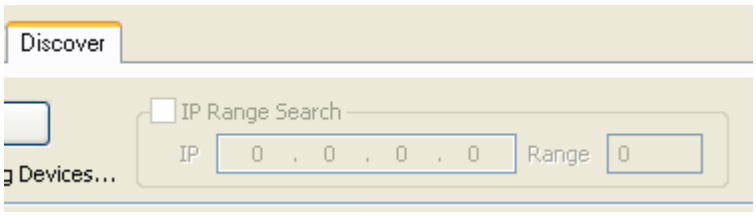
The user can enter the IP address or can select an index from the Phone Book. The encoder can be configured from this dialog as well.

It is also possible to discover automatically all Prodys IP codecs connected to the IP network: In order to make easy the dialling operation over IP networks, a new facility was included in the Dial window. This new tool allows the discovering of the Prodys IP units connected to the network.

There are two options to discover the units on the network:

1.- The PortaNet sends a broadcast message. This is the default option and it is not necessary to enable anything but **the network must support broadcast traffic.**

2.- When the broadcast traffic is not allowed over the IP network still it is possible to discover the units on the network. The "IP range search" checkbox must be enabled and a range of IP addresses must be introduced for a quicker search.



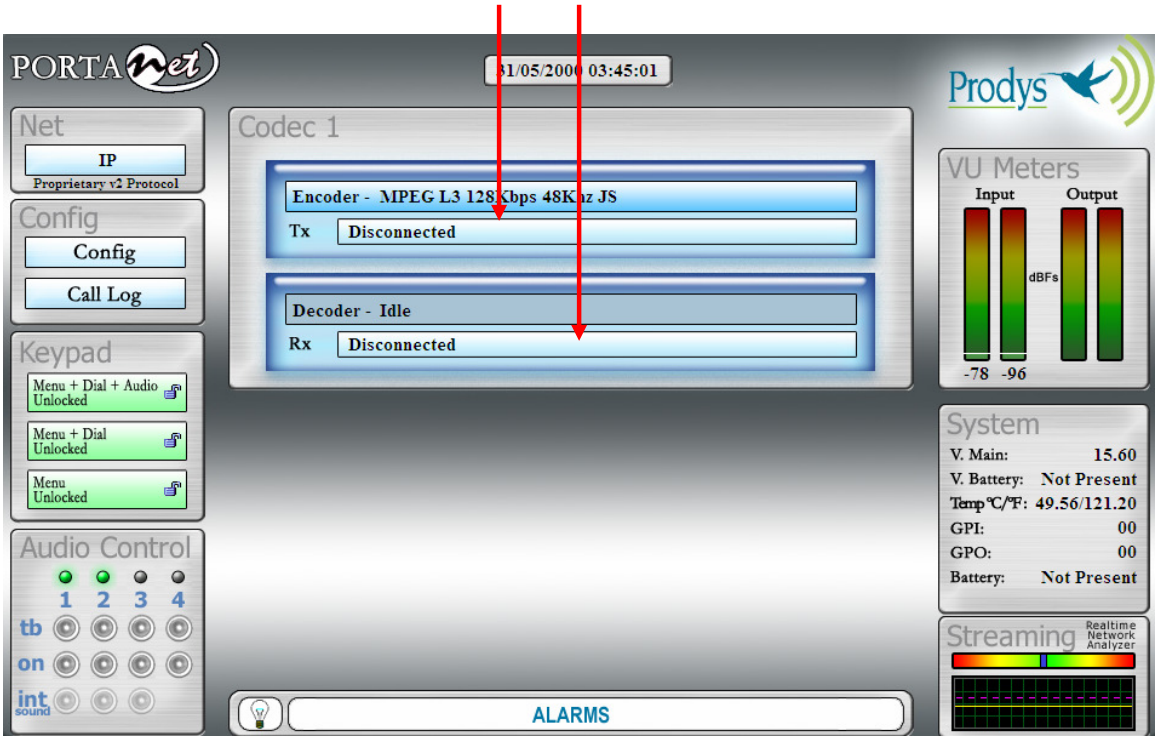
After selecting the address, to make the call just click on "Dial".

VIII.7.4 Disconnecting the Line

By clicking on the Tx or Rx Connection areas.

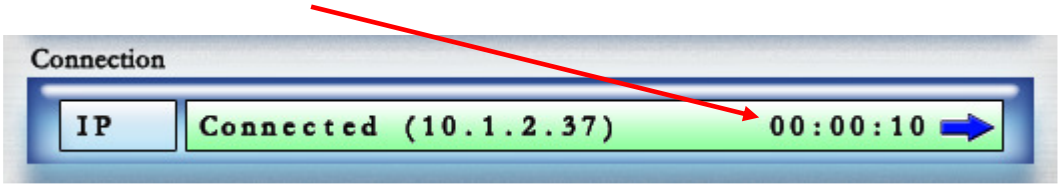
VIII.7.5 Line Status

The Line status is shown on the Tx and Rx Connection areas:



These are the different status messages which can be displayed:

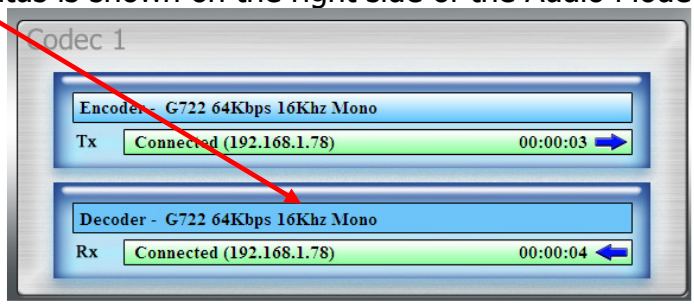
- "No physical Line": The communication line is not physically detected. Most likely the interface is not plugged in. The Display shows "DOWN".
- "Connected": The line is connected. An arrow will indicate if it is an incoming or outgoing call.
  The duration of the call also is indicated.



- "Disconnected": The line is detected physically but no connection is being made. Beside the text appear the disconnection codes.
- "Calling": In the process of making a connection.
- "RING": Receiving a call on the line.

VIII.7.6 Decoder Status

The Decoder status is shown on the right side of the Audio Mode area:

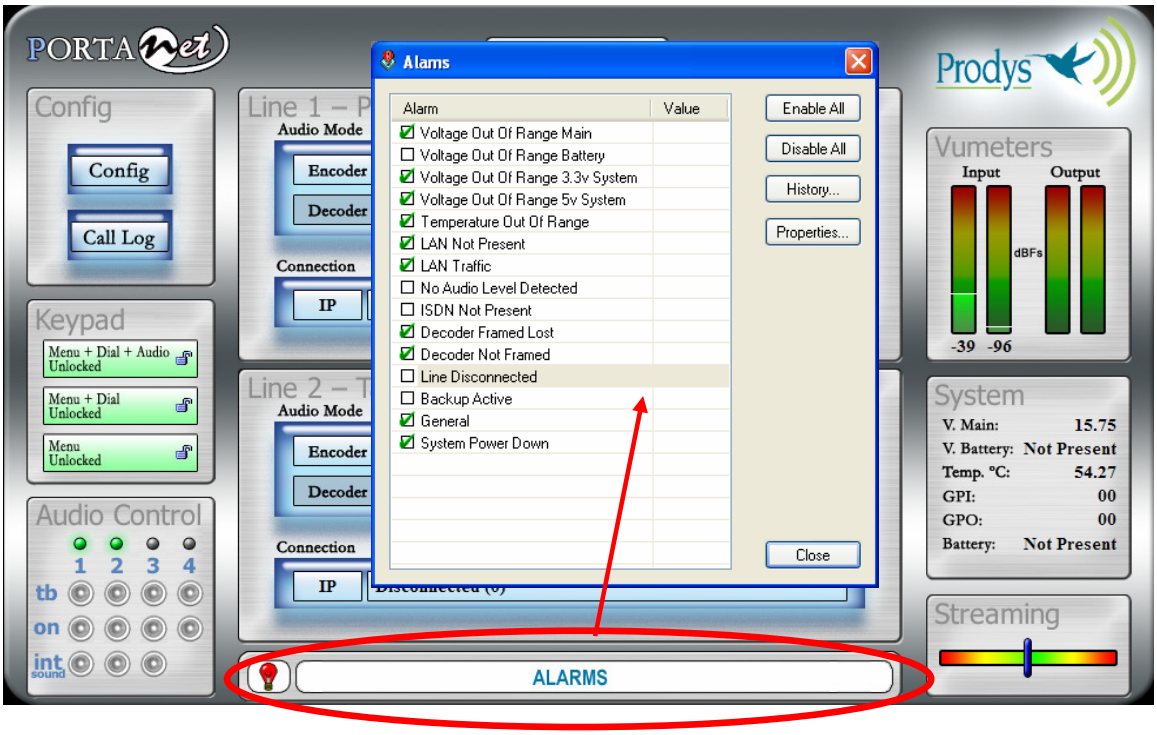


The Decoder status:

- "Searching": The decoder is not synchronized.
- "Framed" : The decoder is synchronized.
- "Idle": The line is disconnected.
- "Not available": Decoder not available.

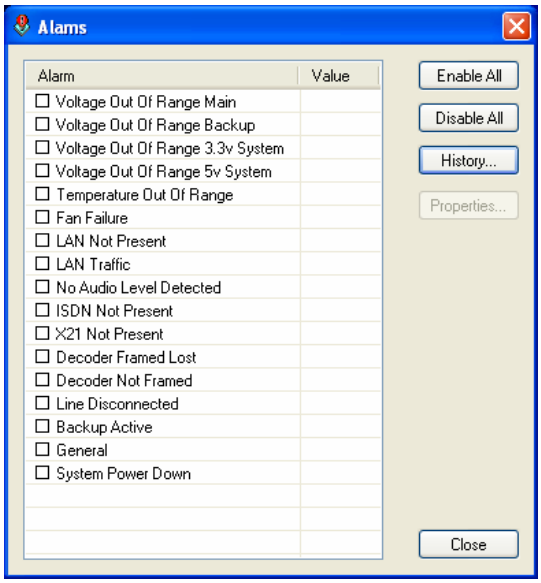
VIII.8 Alarms

The alarm window allows the user to select the alarms that the unit will check. The unit will notify the occurrence of each of the selected alarms. It is possible to configure the unit to send SNMP traps or emails to notify alarms information.

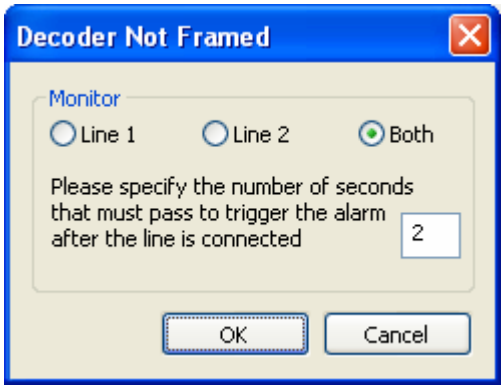


VIII.8.1 Selecting Alarms

The unit supports many different alarms. The selection menu is opened by clicking over the Alarm area.



- **Voltage Out of Range 12 v Main:** The voltage from the main power supply (AC/DC) is out of range.
- **Voltage Out of Range 12 v Backup:** The voltage from the backup power supply is out of range**.**
- **Voltage Out of Range 3.3 v System:** The voltage from the 3.3 DC/DC converter is out of range**.**
- **Voltage Out of Range 5 v System:** The voltage from the 5 DC/DC converter is out of range**.**
- **Temperature Out of Range:** The temperature is out of range (over 50ºC).
- **LAN NOT Present:** Not physical level detected in the LAN port.
- **LAN Traffic:** The LAN traffic is higher than the 90% of the capacity of the network.
- **No Audio input level detected**: No audio is present on the input. A threshold and a time period can be defined to activate the alarm.
- **ISDN NOT Present**: Not physical level detected in the ISDN port.
- **Decoder 1/2 Framed Lost**: The Line is connected but the decoder is not framed, but it was framed before. The alarm will be activated also if the unit is turned off. In this case, the alarm will indicate the date and time when the decoder lost the synchronization, that is, when the unit was turned off. The alarm will be activated until the unit is framed once the line is connected again. If the connection is not possible, the alarm will be marked as finished.
- **Decoder 1/2 Not Framed**: The Line is connected but the decoder is not framed. This new alarm doesn't require that the decoder was framed previously. For this reason, it is necessary to specify a window time to decide when the alarm will be activated once that the NOT FRAMED condition has been detected.



Otherwise, the alarm would come up as soon as the connection is established, given that the Decoder needs some time to detect the incoming compression mode.

- **Line Disconnected**: Line 1/2 has been disconnected unexpectedly, manually disconnected or both.
- **Backup Active**: The ISDN backup line is working.
- **General**: In internal hardware error was detected by the software.

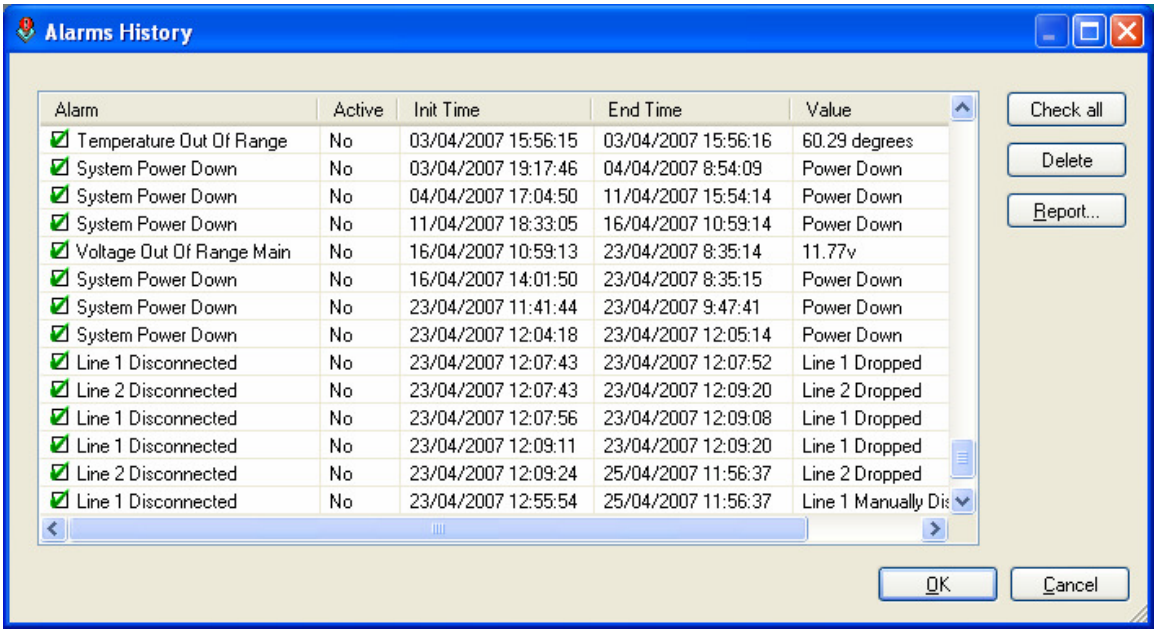
- **System Power Down**: This new alarm allows to know if the unit was turn off and the time that remained in that condition.

VIII.8.2 Monitoring Alarms

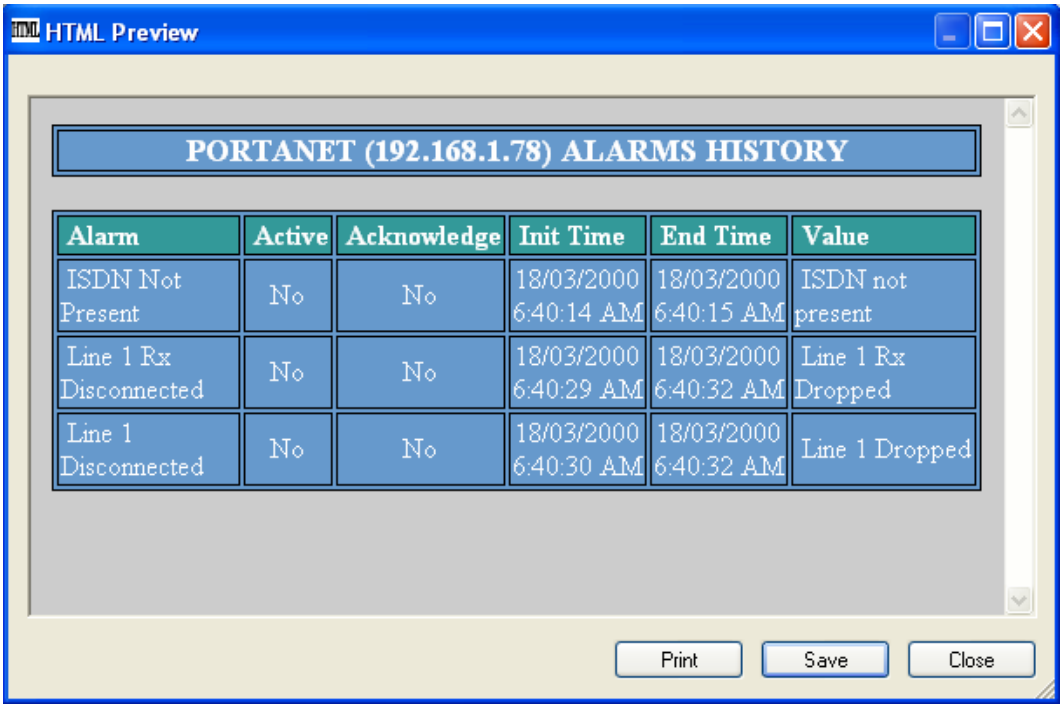
When one of the selected alarms is detected, the alarms window is highlighted

VIII.8.3 Alarms History

The alarms are saved on the non volatile memory of the PortaNet. It means that it is possible to know what happened on the past even if the unit was disconnected of the network. The alarms history can be showed by clicking over the lamp or over "History" option in the alarm selection dialogue.



An historical report of alarms in html allows the user to preview and print alarm events.

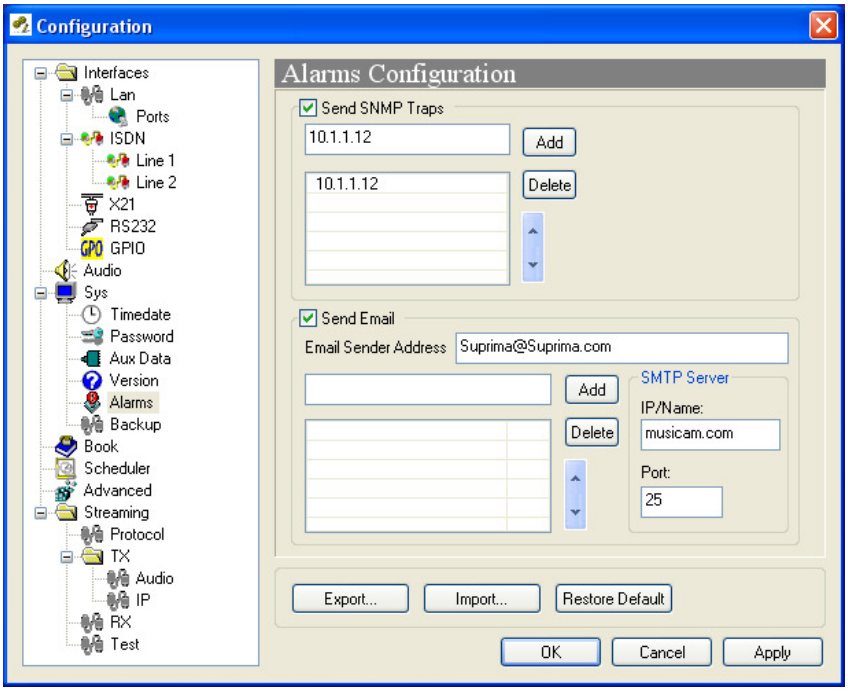


VIII.8.4 Alarms Notification

There are two different ways to notify alarm events: SNMP v1 traps and emails.

VIII.8.4.1 SNMP traps

It is possible to select an IP address where the SNMP traps will be sent. An SNMP manager located at this IP address will receive and process the information according to the SNMP protocol. These SNMP traps will notify alarm information, that is, an SNMP trap will be sent when an alarm is activated or deactivated, with the time and type of alarm. For this to happen, alarms have to be enabled (see chapter VIII.2.2.5 - Alarms. PortaNet can be fully monitored though SNMP protocol.



VIII.8.4.2 Email

From the alarm dialog the user can enter several e-mail addresses where this information will be sent, to inform about the changes in the status of any alarm. It is also possible to define the sender address of the email.

How does the PortaNet work?

Given the different nature of the two communication standards that the PortaNet supports, the unit has to adopt different configurations while adapting to the characteristics of each network. What's more, in certain situations, PortaNet is capable of dealing with two independent communications at the same time. This can be a cause of confusion for the user, who may be expecting a certain behaviour or action from the unit but observes another. This chapter is therefore a practical guide to help in understanding just how the PortaNet unit works under different configurations, especially the more unusual ones.

IX.1 Selecting the communications interface

Firstly, when we refer to the 'communications interface' we are always referring to the communications port that is used for the transmission of 'audio data'. The NET option in the main menu is for setting this up. There are three settings available with the PortaNet:

a. PortaNet operating as "IP CODEC": The PortaNet will be configured to establish communications via the Ethernet port using the IP protocol.
b. PortaNet operating as "ISDN CODEC": The PortaNet will be configured to establish communications via a basic access ISDN connection. The PortaNet has a multi-protocol ISDN terminal socket ready for connection to a basic rate 2B+D interface.

The Ethernet interface is always available for remote control of the unit, even if it is configured to operate as an ISDN codec.

You should first select the network type - i.e. the communications interface - to be used for audio data transmission and reception. The menu will only show the configuration options relevant to the network type selected. Of course, there are a number of general parameters that will be the same across all network types and so will not be affected when the network type is changed.

IX.2 Configuration parameters that are dependant on the network type selected

There are two fundamental reasons for which the user must change the PortaNet configuration parameters when a new communications interface is selected:

- The available bandwidth over the selected network (IP, ISDN). This places limitations on the bit-rate that can be selected and the amount of compression. As an example: over ISDN it is not possible to send uncompressed audio and the maximum bit-rate for any algorithm is 128Kbps.

- The number of available channels: Number of bidirectional communications which can be achieved. Note that over ISDN and IP, up to two independent mono communications can be achieved.

The following table shows what the differences between the various communications networks regarding PortaNet operation:

NET	ALGORITHM	BIT RATES (Kbps)	Nº CHANNELS
IP	PCM	16 bits @ 48 KHz.	2 Mono[13] or 1 Stereo
	G711	64	
	G722	64	
	MPEG1,2 LII	64,128,192,256,384	
	MPEG1,2 LIII	64,128,192,256	
	MPEG AAC LC.LD	64,128,192,256,384	
	MPEG 4 AAC HE	24,32,48,56,64,128	
	aptX™	64,128,192,256,384	
ISDN	G711	64	2 Mono or 1 Stereo
	G722	64	
	MPEG1,2 LII	64,128	
	MPEG1,2 LIII	64,128	
	MPEG 2,4 AAC LC	64,128	
	MPEG 4 AAC LD,HE	64,128	
	aptX™	64	

IX.3 PortaNet working as a "DUAL CODEC" over ISDN

When we select ISDN as the communications interface, it is possible to establish two, totally independent MONO connections for each 64 Kbps B Channel. This means that for each channel or line of communication (Line 1 or Line 2 as seen by the PortaNet) it is possible to send an audio signal encoded with any of the available codec algorithms. The menu offers the option to configure Encoder 1 and Encoder 2 separately with MONO signals on each channel. To avoid incorrect configuration there are therefore certain restrictions while working with the PortaNet as a "DUAL CODEC":

[13] Over IP, the second MONO communication is fixed to G722.

- If Encoder 1 is configured in any mode other than DUAL or STEREO/JSTEREO[14] the Encoder 2 option is disabled and the PortaNet will not work as a "DUAL CODEC".

The Encoder 2 is not available also when NET = ISDN and bit rate = 128 Kbps.

- Encoder 2 can only be configured to work in MONO.

IX.4 PortaNet working as a "DUAL CODEC" over IP[15]

There are two different operation modes: SIMPLE and DOUBLE mode. In DOUBLE mode, the unit will work as usual, that means, it can establish two independent MONO communications via IP. In SIMPLE mode, the unit can establish only one MONO or STEREO communications via IP.

In DOUBLE mode over IP, the following restrictions will be applied:

1.- Encoder 1 and Encoder 2 can only be set in MONO.
2.- Encoder 2 (assigned to line 2) can only be set in G722.

This configuration can be set both from the menu and from the Web Page.

[14] PortaNet is aimed to be used as a MONO audio codec. To configure the unit to be used in STEREO, please read the chapter "Configuration Examples".

[15] Operation mode only available when selecting Prodys Proprietary Protocols over IP.

IX.5 About how the Decoder works and automatic searching

The PortaNet Decoder system doesn't need to be configured as it is totally automatic. However, it is necessary to point out some aspects of how it works as it depends on the format of incoming data. The PortaNet is able to synchronise automatically to the following algorithms under the following conditions:

Algorithm	**IP Codec (NET = IP)**	**ISDN Codec (NET = ISDN)**
G711	YES	YES
		The system detects incoming calls such as voice calls and configures the Encoder and Decoder to G711.
G722	YES	YES
		Needs G722 encoded audio since it is synchronised by means of statistical framework. **It does not work if encoder is configured to aptX , AAC HE or J52.**
MPEG1,2 LII MPEG1,2 LIII MPEG2,4 AAC LC MPEG 4 AAC LD	YES	YES
		It does not work if encoder is configured to aptX , AAC HE or J52.
Mpeg AAC HE & aptX™	YES	NO
	aptX not with SIP/SAP as IP protocols.	It is necessary that the Encoder is also configured to AAC or aptX.
PCM Linear Audio	YES	NO
		Not Available.

Taking into account the previous table the user must also bear in mind the following:

1. Over ISDN, it is only possible to set different encoding and decoding algorithms independently, with those algorithms that support automatic synchronising. For example, it is possible to transmit on G722 and receive on MPEG Layer II.

Although G711 is detected automatically in ISDN mode, the Encoder is also configured automatically because it is not possible to use it in any other combination.

2. If we want to receive an algorithm that does not support automatic synchronisation it is necessary to configure the Encoder to the same algorithm.

3. If we want the Encoder to switch to the algorithm detected by the Decoder, the Encoder should simply be set to AUTO mode.

IX.6 The PortaNet operating as IP codec (Proprietary Protocols)

The operation of the PortaNet as an IP codec offers three operational modes: UNICAST, MULTICAST and MULTIUNICAST[16].

IX.6.1 UNICAST communications

The term UNICAST is used in the networking world to refer to the connection to a single destination. Applied to the PortaNet, this is when a point-to-point connection is created between two units bi-directionally.

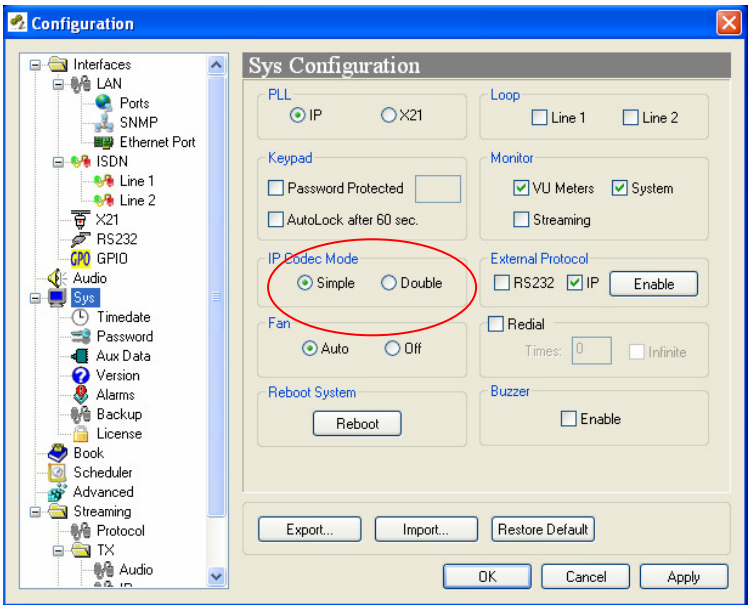
IX.6.2 Using line 2

There are two different operation modes: SIMPLE and DOUBLE mode. In DOUBLE mode, the unit will work as usual, that means, it can establish two independent MONO communications via IP. In SIMPLE mode, the unit can establish only one MONO or STEREO communications via IP.

In DOUBLE mode over IP, the following restrictions will be applied:

1.- Encoder 1 and Encoder 2 can only be set in MONO.
2.- Encoder 2 (assigned to line 2) can only be set in G722.

This configuration can be set both from the menu and from the Web Page.

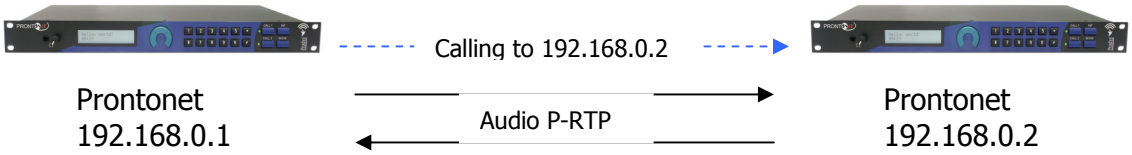


[16] MULTIUNICAST is available from version 5.2.1 onwards.

IX.6.3 Establishing a UNICAST connection from the PortaNet

The procedure for establishing a connection is very similar to that of making an ISDN call; we enter the call initiation menu by pressing the CALL 1 key and enter the IP address of the unit we wish to connect to. As we have already mentioned, the audio data connection is bi-directional (as in an ISDN codec) and we need two connections, one in each direction. For this, the PortaNet that receives and accepts the call will automatically call back to the originating PortaNet and establish a reverse connection.

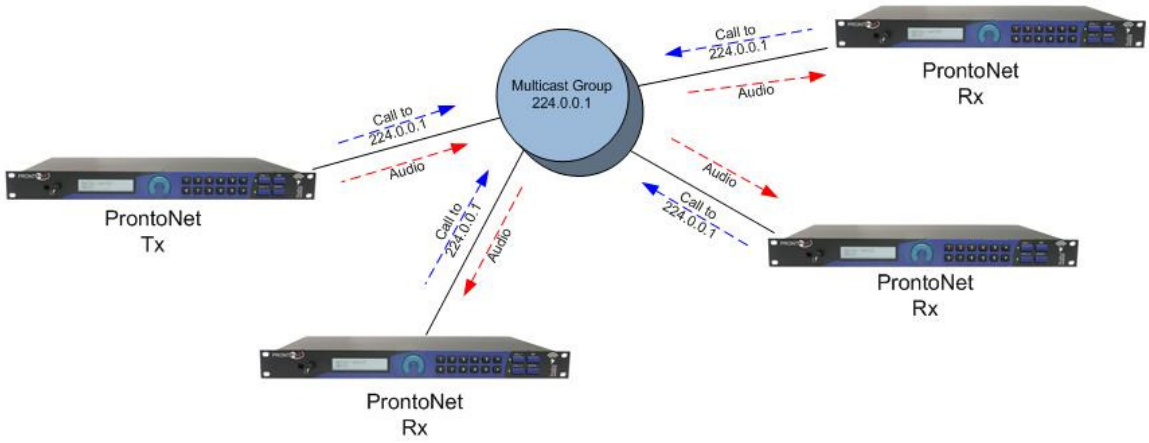
This is all represented in the following diagram:



For more information about which protocols Prodys IP units use, please read chapter IX.6.6.

IX.6.4 Establishing a MULTICAST communication from the PortaNet

With MULTICAST the calls must be made from both ends. Both the sender of the data and all the receivers of the data must call to establish a connection to the multicast group. The multicast operation can be shown in the following diagram:



To initiate audio distribution the role of the transmitter unit and the roles of all the receiver units must be set up for the group. These modes can be selected from the menu **CONF/PORTS/LAN/MULTICAST** where you select **Tx** (transmitter) or **Rx** (receiver). Once this is done, the calls can be established. This can be done in any order, that is, calls can be set up first from the transmitter and then from each of the receivers, or the other way round. The

important thing to note is that when the transmitter is connected to the group, it will start the streaming of audio data immediately. If there is no receiver connected, the audio will simply not be received. As soon as receivers are connected to the multicast group they will receive the audio data that is being streamed. Equally, if the receiver is connected to a multicast group where the transmitter is not operating yet, the audio output is muted.

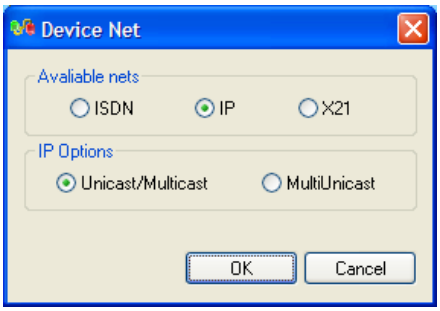
- **Multicast considerations:**

  - Internet Protocol (IP) multicast is a bandwidth-conserving technology that reduces traffic by simultaneously delivering a single stream of information to thousands of corporate recipients and homes.
  - Multicast is based on the concept of a group. An arbitrary group of receivers expresses an interest in receiving a particular data stream. This group does not have any physical or geographical boundaries—the hosts can be located anywhere on the Internet. Hosts that are interested in receiving data flowing to a particular group must join the group using IGMP. All this is done automatically by PortaNet when establishing a connection.
  - Multicast traffic is rejected when going through the Internet, since most IP servers on the Internet do not currently support the multicasting part of the protocol, except when using VPNs, because VPNs encapsulates IP packets as unicast frames, so routers simply see an ordinary packet.
  - All IP multicast group addresses will fall in the range of 224.0.0.0 to 239.255.255.255, but some of them are reserved, that's why the range of addresses from 224.0.1.0 through 238.255.255.255 are called globally scoped addresses.
  - There should only ever be one transmitter connected to a MULTICAST group or else audio reception errors will occur.
  - For transmitting MULTICAST audio the Prodys proprietary protocol "Prodys eXtended Real Time Protocol (PX-RTP) or SAP (according to the new standard for IP compatibility) will be used.
  - The PortaNet transmitter cannot be in automatic encoding mode.
  - To guarantee a constant delay all the units must synchronise their clocks. Each receiver will activate a clock-sync algorithm that adjusts its PLL (Phase-Lock Loop).

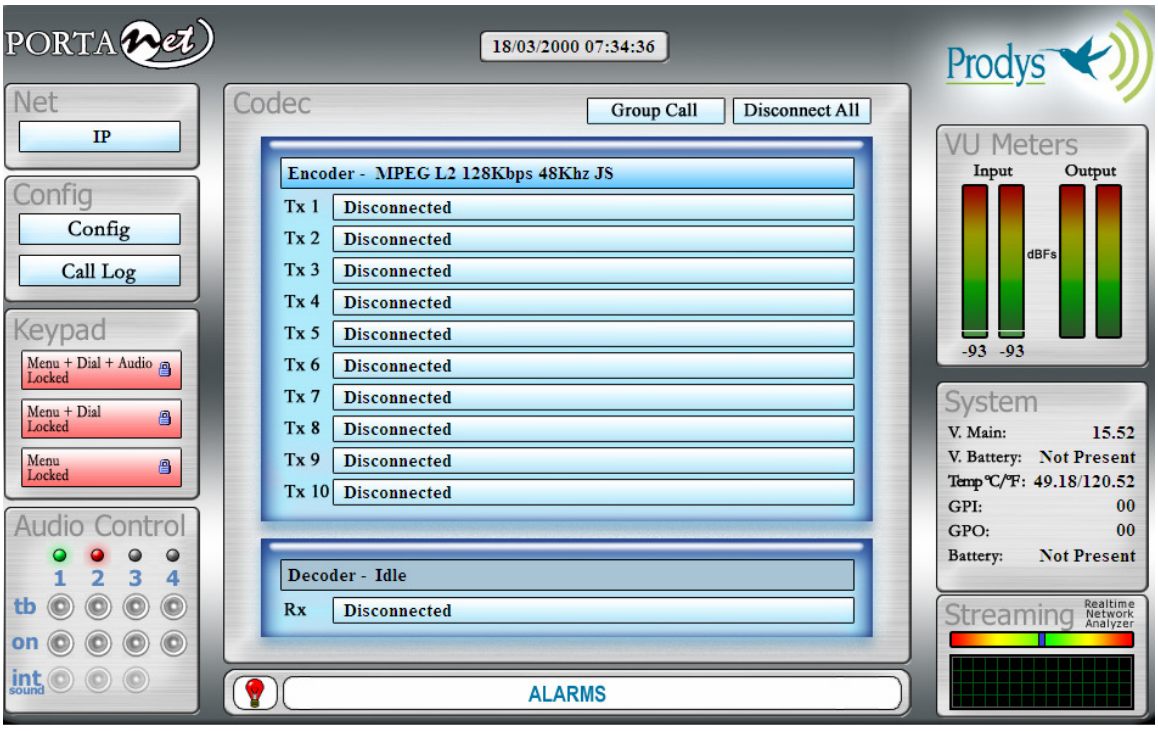
IX.6.5 MULTI-UNICAST

MULTI-UNICAST allows the user to establish point to multi-point connections by means of unicast connections. The same audio (one encoder) can be sent up to 10 different locations concurrently. This functionality overcomes the lack of Multicast support on some networks.

To select this new feature, open the new 'NET' menu by clicking on the 'NET' button at the top on the left in the web page.



The main window will show all the available connections:



Depending on the compression mode, up to 10 Tx connections will be available:

- MPEG L2, L3 y AAC: Up to 10 Tx + 1 Rx.
- PCM, G711, G722, APTX: Up to 3 Tx + 1 Rx.

As can be seen, there is an independent control bar for each connection. Each of the connection bars will show the line status in real time. The procedure to establish a call is the same as that for Unicast connections.

As for Unicast connections, the Tx and Rx parts of the connection are divided so that it is possible to receive from one end at the same time the unit is sending audio on one or more Tx connections.

In addition, it is possible to make or hang up several connections at the same time by using the 'Group Call' and 'Disconnect All' buttons respectively.

This operation mode is compatible neither with the IP 'DOUBLE MODE', nor with BackUp operation mode.

IX.6.6 Prodys Proprietary set of protocols

Prodys developed this proprietary set of protocols to carry out IP streaming connections, due to the lack of a standard in this regard:

- Prodys Real Time Control Protocol (P-RTCP): This is a protocol based on TCP that allows for the establishment and termination of a connection as well as for the negotiation of the codec mode (automatic audio synchronisation in all modes).
- Prodys Real Time Protocol (P-RTP): This is a protocol based on UDP used for the transmission of audio.
- Prodys eXtended Real Time Protocol (PX-RTP). This is a protocol based on UDP used for the transmission of multicast audio.
- Prodys Upgrading/Identifying Protocol. This protocol is based on UDP and used to identify/upgrade the units.
- Prodys External Protocol (P-XP). This protocol is based on TCP and can be used for controlling the units from an application other than the web page or ProdysControl.
- Prodys Auxiliary Data Protocol (P-AUXP). This protocol is based on UDP and used for transmitting/receiving auxiliary data[17].
- Prodys U-bit Protocol (P-UbP). This protocol is based on UDP and used for transmitting/receiving the User Bit from the AES/EBU frame. Available from version 4.8.0 on.

It is worth mentioning here, that at the time of writing, Prodys has a close relationship with the EBU group N\ACIP which have started the first committee in order to define an standard in audio contribution over IP. Prodys work in the definition of the standard along with the members of the EBU group, in

[17] Available from version 4.8.0 onwards.

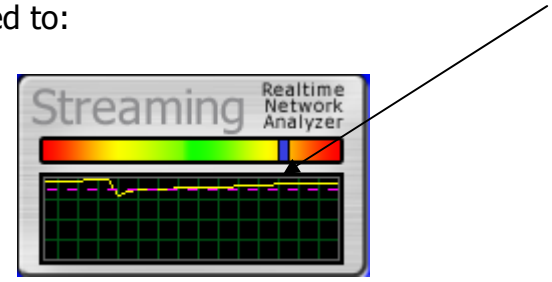
accordance with the proposals from the radio stations involved in this project.

IX.6.7 Proprietary (set of protocols) v2[18]

This set of protocols is based on the previous one, and it uses the same ports and different protocols. But, with this new protocol, it is possible to obtain network related parameters such as jitter, lost or disordered packets in real time during the audio connection. To be able to get all this information, Proprietary Protocol V2 should be configured as audio protocol. This protocol is not compatible with the previous one (version 1). Proprietary protocol version 1 will allow the user to get information about jitter and buffer usage, but will not allow the user to obtain information about lost and disordered packets.

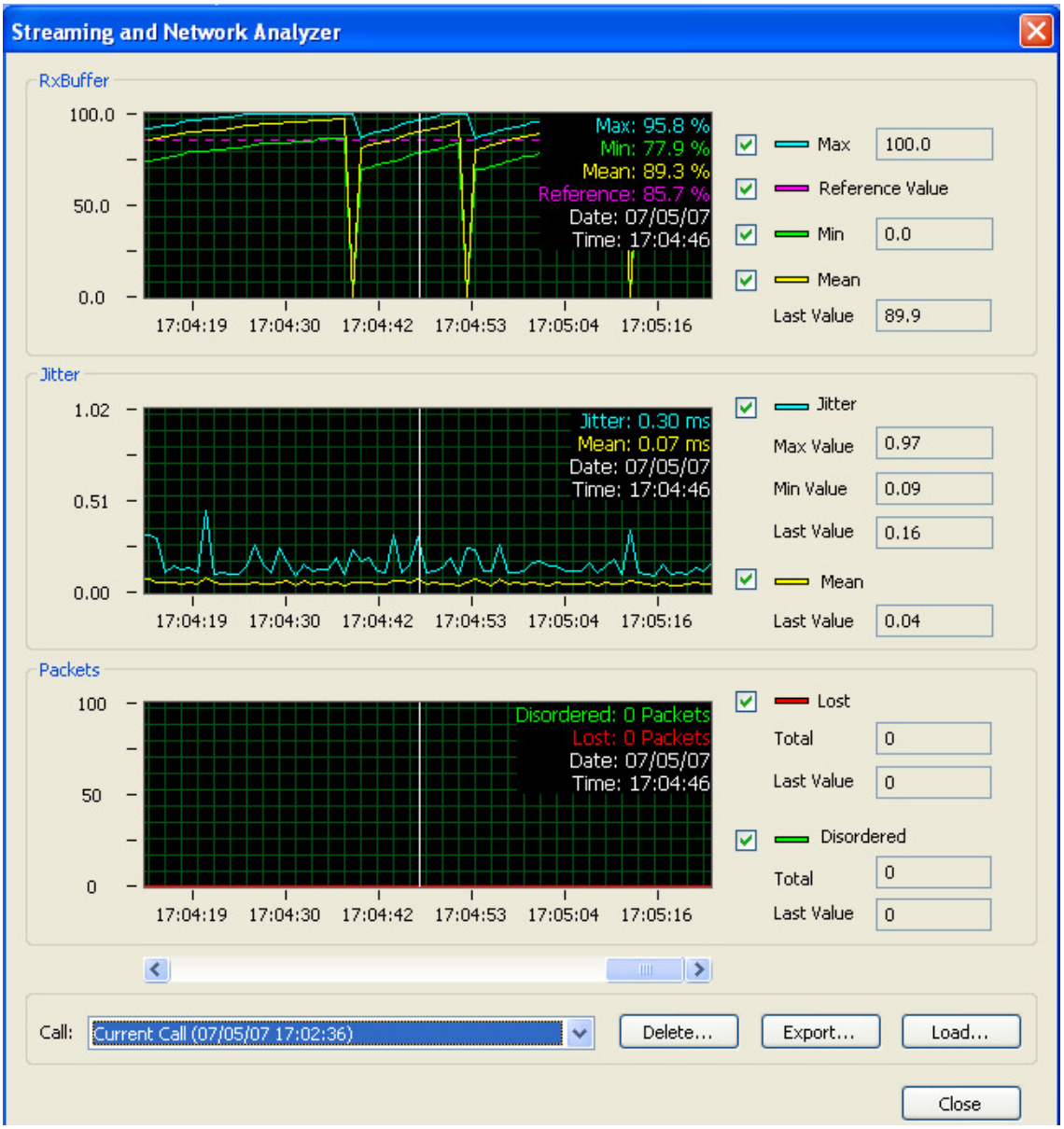
This information will be saved separately for each connection in RAM memory. Up to 24 hours of data from different calls can be stored.

Once the connection is established, the user can access the 'real time network analyzer' by clicking on the 'buffer occupation graph', in order to get information related to:



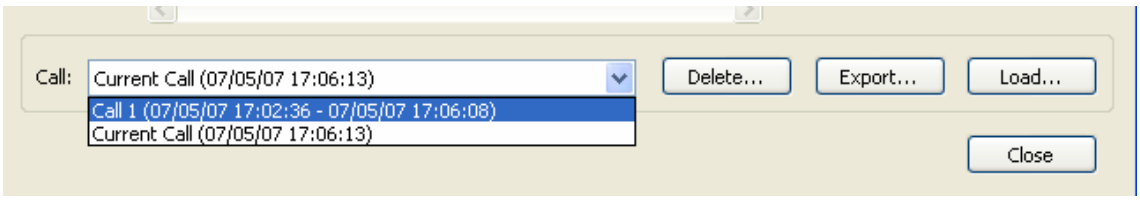
-Rx occupation: With average, maximum and minimum usage. Very low percentage of buffer occupation will cause audio interruptions and drop-outs.
-Jitter: High jitter values will match with low buffer occupation.
-Lost and disordered packets.

[18] This protocol is available from version 5.2.1 onwards.



All the information is displayed in different graphs, synchronized to each other, so that the user can move through all the data very easily.

Data from each connection is stored independently so that it is possible to access data from connections other than the current one. In addition, it is possible to delete, export or import data from any previous call.



IX.6.8 PRODYS PORTS for Prodys Proprietary protocols (v1 & v2)

These are the **default** ports used by Prodys IP Codecs for their IP connections:

HeraFlash &
Prodys Control
UDP:50013

Web Page
TCP 80: HTTP
TCP 50011: Web Page
TCP 50017: ProdysControl

Audio Streaming
TCP 50019: P-RTCP (Control)
UDP 50021: P-RTP Unicast L1
UDP 50023: P-RTP Unicast L2
UDP 50025: PX-RTP Multicast L1
UDP 50027: PX-RTP Multicast L2
UDP 50037: P-AUXP Auxiliary data (from version 4.8.0 on)
UDP 50039: P-UbP U-BITs (from version 4.8.0 on)

Test Streaming Tool
TCP 50033
UDP 50033

External Protocol (P-XP)
TCP 50031: Control Port
TCP 50035: Status Port

It is possible to change these ports from the web configuration window. Please, for more information about port changing, read chapter VIII.2.1.1.1 – Ports.

IX.7 SIP

The European Broadcast Union (EBU) is promoting the interoperability of audio codecs for any manufacturer. For this purpose the use and the application of a subset of the Internet Protocols has been proposed. This effort will allow to setup in a friendly way audio streaming communications between several vendors of equipment.

The deployment of SIP Protocol Servers among the network will support calling remote parties just by invoking their network name regardless of the actual

public o private IP addresses. Neither is required to manage the office routers and the firewall's TCP/IP ports for any new communication path.

The physical location, often related to fixed IP addresses or subnets, is not further meaningful. In this sense, specially portable audio codecs will profit with an easy call procedure. Neither is required to agree ahead on the audio compression type and data stream rate, because the SIP Protocol manages by itself to negotiate the convenient communication details with the remote party.

For Prodys' former customers performing a call using SIP is much the same as using Prodys' proprietary protocols given that the SIP configuration is set.

SIP supports up to date only unicast streaming. Any PortaNet's audio compression mode is supported also if SIP protocol is in use.[19] For further details please refer to the "Technical Description" at this document.

- **SIP Port**: TCP/IP port dedicated to SIP signalling for establishing, updating or finishing a call. Port 5060 is mandatory by the standard if direct calls without gateways are expected.

- **Audio Port**: TCP/IP port dedicated to the RTP communication, this is the actual port for the audio streaming payload. By default the number 5004 is reserved.

If SIP server support is granted by your network:

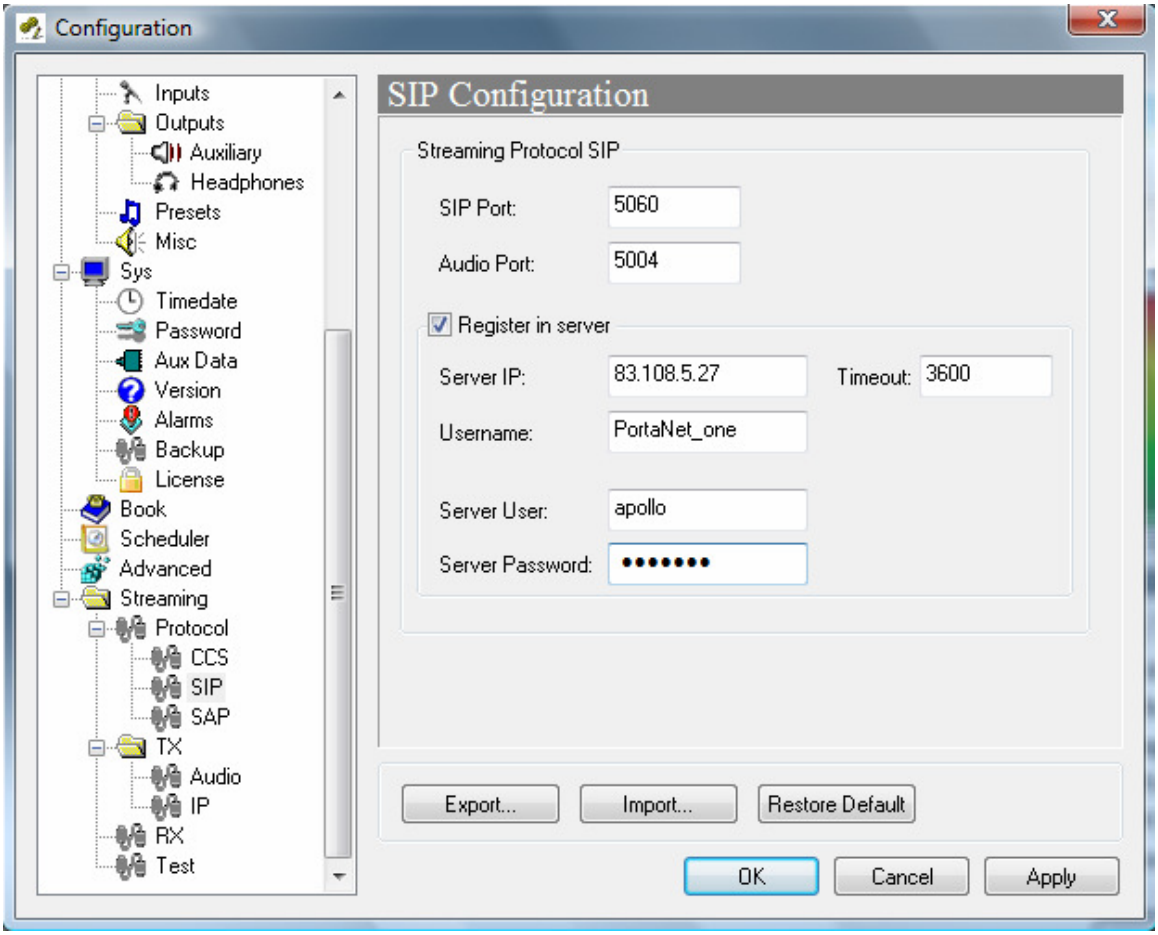
- **Server IP**: IP address of the SIP Server. At this point is registered periodically the proper user information for full protocol support.

- **Timeout**: Time in seconds before the user information registered at the previous server is flushed. Therefore the information about the SIP user are updated from time to time.

- **Username**: Your alias on the Internet regardless of your current IP address. By this alias you are identified for other SIP participants.

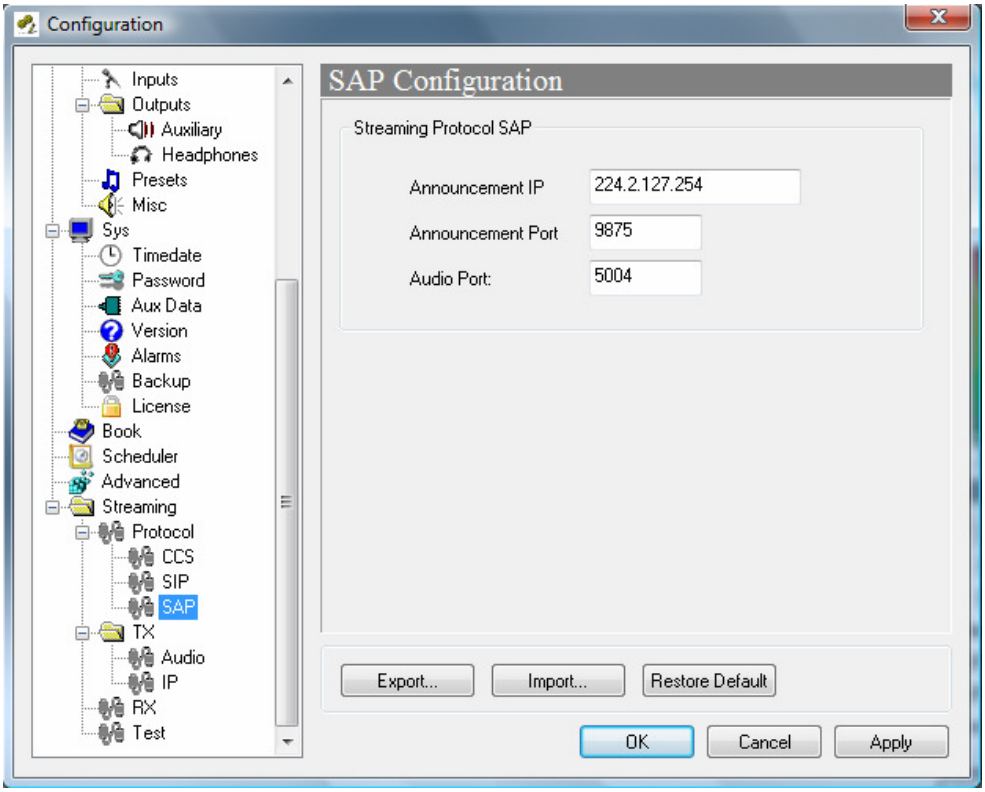
- **Server user & password**: Some SIP servers requires authentication before proceeding the register of SIP users.

[19] The apt-X compression is still undefined by the EBU Tech 3326 standard. For compability reason this mode is not supported if SIP protocol is selected.



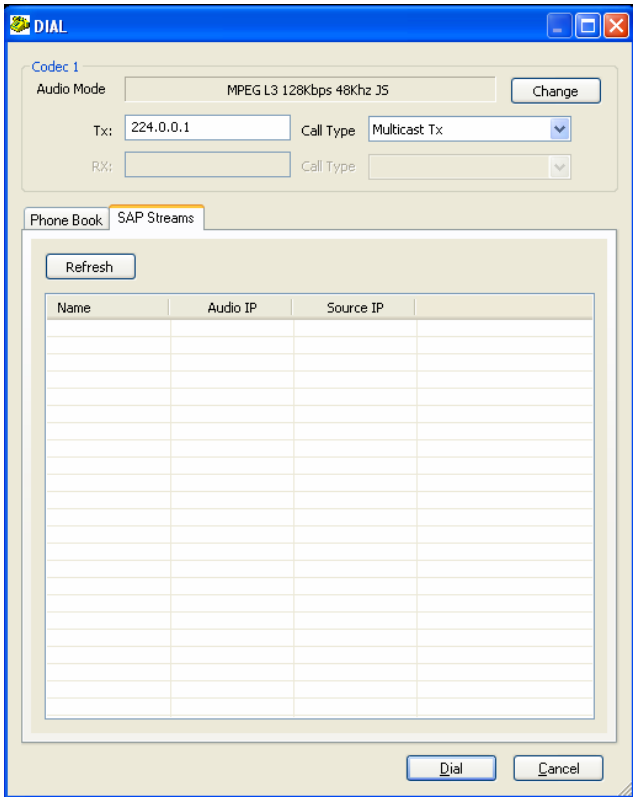
IX.8 SAP

When selecting SAP as communication protocol, only multicast calls will be available (point-to-multipoint). In addition, the user will have to indicate the announcement IP address and port on which the audio streaming is 'announced'.

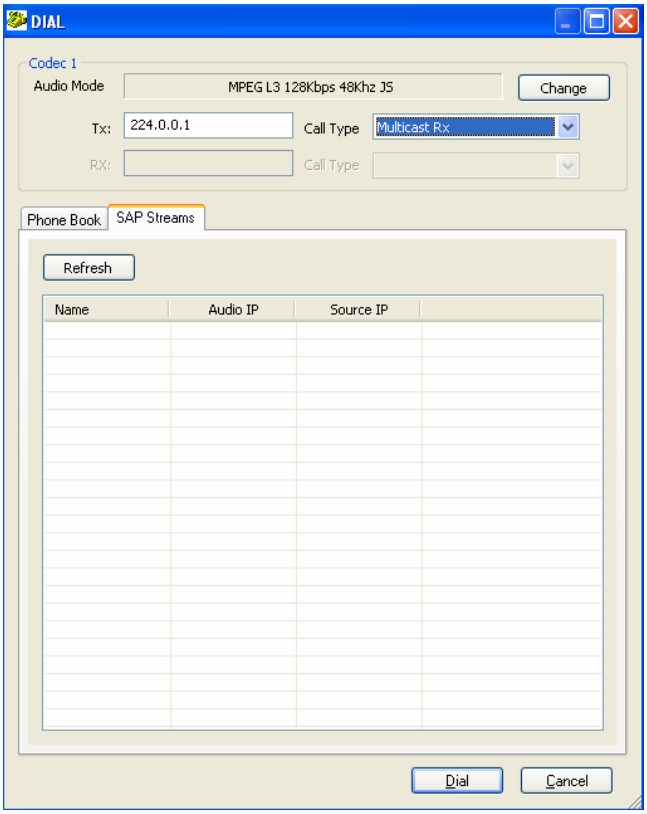


The default values for these parameters are the default ones defined by the SAP standard (RFC 2974). This address and port will be used by the transmitter to 'advert' its audio streaming broadcasting. In the case that these parameters are modified by the user, they should be configured to the same values at both ends, the sender and the receiver.

When making a call with SAP as communication protocol, only Multicast Tx and Multicast Rx call types will be available. To launch an audio streaming over IP with SAP the user shall select Multicast Tx as call type from the dialing window, and type in the multicast IP address to send the audio to.



To receive this audio streaming, the user shall select the Multicast Rx call type, and will introduce the same Multicast IP address as the transmitter.



As many receivers as required can join this multicast address and receive the audio, without any increment in the required bandwidth, given that this communication is based on IP multicast technology.

IX.9 PortaNet operating as an ISDN codec

The PortaNet as ISDN codec is very similar to the Pronto 3, therefore those users that are familiar with the Pronto 3 will find it easy to operate the PortaNet over ISDN.

The most important points that a user must have in mind for this mode are as follows:

1. **ISDN interface:** The ISDN interface of the PortaNet allows connection to a basic rate ISDN line. This interface allows the user two bi-directional channels with a bandwidth of 64 Kbps on each direction. There is an additional channel of 16 Kbps that is used for signalling. This is why a basic rate connection is sometimes represented as 2B+D. The PortaNet identifies each B channel as Line 1 and Line 2 respectively.

   The PortaNet can operate as "DUAL CODEC", which means it can establish two totally independent MONO communications via each B channel at 64 Kbps. In this situation the PortaNet will be able to use two Encoders and two Decoders independently.

   PortaNet is aimed to work as a MONO codec. To configure the unit to work as a STEREO codec please read the chapter XI - Operation Examples.

4. **Communications at 128 Kbps – Inverse Multiplex:**

Certain algorithms permit the compression of the signal at bit-rates other than 64Kbps, for example at 128Kbps. With the PortaNet it is possible to send or receive audio compressed at 128 Kbps using both the B channels of the basic rate ISDN connection. This is done by using 'inverse multiplex' techniques that split the transmission of the full 128 Kbps over two channels of 64 Kbps and then sum the parts together again at the other end.

Inverse Multiplex is often known as 'bonding'. PortaNet supports CCS, Telos and J52 bonding algorithms over ISDN.

5. **Communications combining 128 and 64 Kbps**: It is possible to transmit encoding audio at 128 Kbps and to receive decoding audio at 64 Kbps or vice versa. Here below there are different examples of it.

IX.9.1 Establishing ISDN calls

To make a call the CALL 1 and CALL 2 keys are used, or an entry from the Address Book is selected using the BOOK key. Bear in mind that the use of an entry in the Address Book will also reconfigure the Encoder and so is not limited to only making calls. We need to distinguish between calls at 64 Kbps and calls at 128 Kbps:

- Making calls at 64 Kbps

The CALL 1 key is for calling on Line 1 and the CALL 2 key is for calling on Line 2. On pressing the CALL 1 key the display prompts you to enter a number. Once the number is entered and the ENTER key is pressed, the PortaNet will go ahead and make the call.
The CALL 2 key operates in exactly the same way.

- Making calls at 128 Kbps

Each line needs to be connected independently using the CALL 1 and CALL 2 keys.

To disconnect a call press and hold the appropriate CALL key for more than one second.

IX.9.2 Receiving calls via ISDN

In general you must bear in mind the following when dealing with incoming ISDN calls:

- **Automatic or manual response:** Incoming calls can be answered automatically or manually depending how the menu option ANS is configured for the ISDN port. If the ANS mode is set to manual, the appropriate CALL key must be pressed to accept the call and connect to it.

- **Call filters:** It is possible to record up to three numbers for each line that work as call filters, meaning that the line will only connect to calls that come from these pre-programmed numbers. This option is found in the ISDN set up menus under CNUM (Calling Number).

- **Local number**: It is also possible to assign a single number to each line in a way that the line will only respond to calls to this local number. This can be used if you need to map an ISDN directory number to a specific audio port. This option is found in the ISDN set up menus under LNUM (Local Number).

- **Receiving calls over G711 (Voice mode)**: the ISDN indicates calls on G711 as voice mode. The PortaNet recognises this type of incoming call and proceeds automatically to reconfigure the Encoder and Decoder to this mode. As such, the user need not configure anything to allow the automatic reception of incoming calls using G711.

The Law is selected automatically, depending on the ISDN protocol selected – A Law for Euro ISDN and U-Law for all others.

IX.9.3 Restrictions in ISDN communications

As we have already stated, the PortaNet can work as a DUAL CODEC over ISDN, meaning it can operate with two independent lines. However, this is only possible when two MONO communications are used. The use of Dual, Joint Stereo or Stereo modes creates certain restrictions on the system that are in place to avoid conflicting situations.

1. DUAL, JOINT STEREO or STEREO modes are only available on Encoder 1.

2. When Encoder 1 is configured to DUAL, JOINT STEREO or STEREO, Encoder 2 is not available.

3. **The PortaNet will connect the line but the audio will not be decoded in the following cases**:

   a. The line 2 is connected and the Decoder 2 is FRAMED. PortaNet receives a call in the line 1 and detects that the audio is encoded in DUAL, JOINT STEREO or STEREO mode at 64 Kbps.

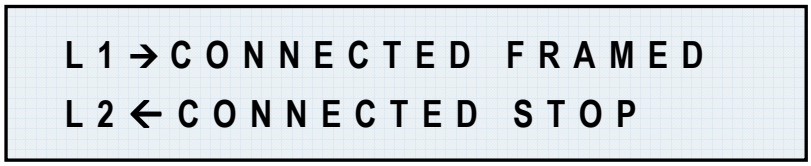
      Line 1 will connect but the audio will not be decoded since PortaNet working as a DUAL CODEC only allows two MONO communications. The display shows an error code to indicate the situation.

   b. PortaNet receives a call in the line 2 and detects that the audio is encoded in DUAL, JOINT STEREO or STEREO mode at 64 Kbps.

      Line 2 will connect but the audio will still not be decoded since these modes are not allowed on Line 2. The display shows an error code to indicate the situation.

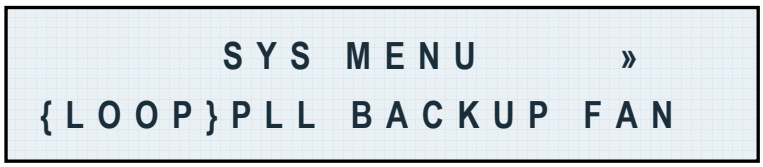
   c. The line 1 is connected and the Decoder 1 is FRAMED in DUAL, JOINT STEREO or STEREO mode at 64 Kbps. PortaNet receives a call in the line 2.

Line 2 will connect but the audio will still not be decoded since these modes are not allowed on Line 2. The display shows "**STOP**" to indicate the situation.



IX.10 How the backup mode works

PortaNet when is configured as IP codec (NET = IP) can use the ISDN connection as backup line, that is, the ISDN will be an alternative communication line when the IP line is dropped.
This option is available in the SYS menu:



IX.10.1 MASTER & SLAVE Configuration

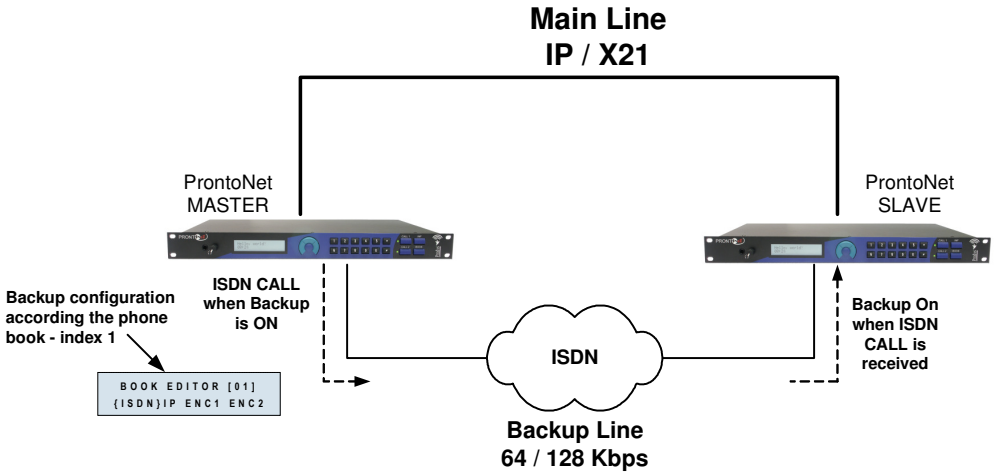
The backup operation is based on the interaction between two PortaNets, one of them working in MASTER mode and the other end configured as SLAVE. The user must configure each unit in one of these modes before to enable the backup facility. The option SYS – BACKUP - MODE allows the selection of Master or Slave.

Once the backup option has been validated, the PortaNet configured as MASTER monitors continuously if the main line is working appropriately in order to decide if the main line is dropped or, in case the unit is working in backup mode (ISDN connection), if the main line (IP) has been re-established.

The PortaNet MASTER uses the information stored in the index 1 of the phone book to configure the unit and to call to the stored numbers.

The Pronto Backup SLAVE only works in backup mode depending on if there is communication by the ISDN line.

Please, find below an example of PortaNet backup application to note the differences between both working modes.



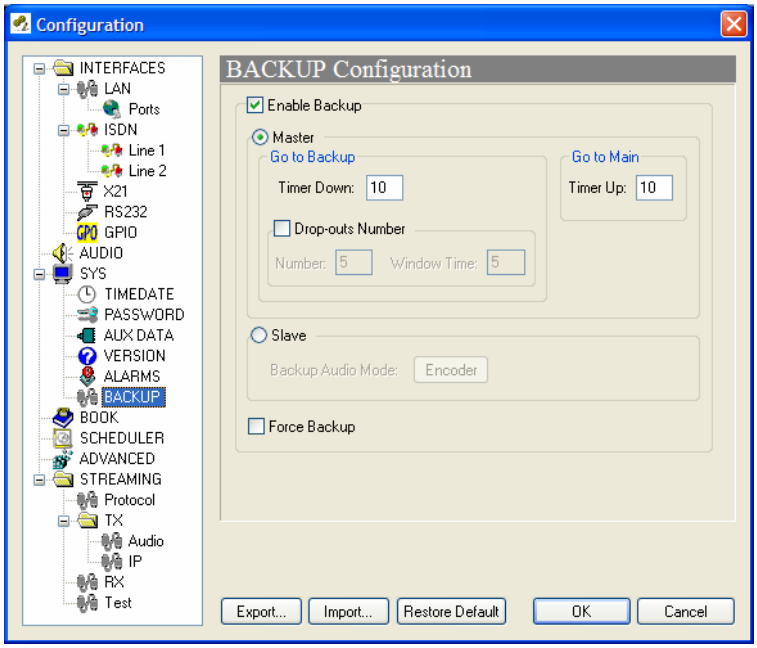
As we can see, the PortaNet configured as MASTER monitors the main line and it decides when the backup ISDN line must start to work. The PortaNet SLAVE only works in ISDN depending on if there is communication by it.

The user must be careful to disable the backup mode (BACKUP_ENABLE_OFF) to avoid that there was any interference in the installation and configuration operations of the unit.

IX.10.1.1 PortaNet MASTER operation

As we mentioned before, the PortaNet configured as MASTER monitors continuously if the main line is working appropriately once the backup option has been validated in the menu (BACKUP-ENABLE-ON). The PortaNet MASTER decides if the main line is dropped when the audio synchronism has been lost for the programmed time in TIMER-DOWN option of the menu or when as many drop-outs as defined in the Drop-outs option of the menu have occurred within the period of time configured by the user[20].

[20] This criteria for the BackUp operation was included on version 4.8.1.



When the audio synchronism has been lost during the programmed time by the user (TIMER-DOWN), or there have been as many drop-outs as defined by the user within the "window time", the unit will be configured in backup mode according the encoding configuration stored in the index 1 of the phone book. Once it has been configured, the codec will proceed to call to the programmed numbers in the same index the number of calls (one or two B channels) will depend on the mode that the user has chosen (64 o 128 Kbps) when the unit starts working in backup mode.

The PortaNet will try the connection as many times as it will be necessary until the connection will be established or until it decides that the main line has been recovered. If the backup is disabled during the calling process or when the ISDN communication is connected, the unit will proceed to work at IP or X21 mode independently if the main line is recovered or not.

While the unit is working in backup mode, the unit is monitoring the main line in order to decide if the line has been recovered or not. If the audio synchronism is detected in the received audio by the main line, the programmed countdown in the TIMER-UP starts. If the countdown finishes, the PortaNet goes to IP mode again and therefore leaving the backup mode after ending the communication by the ISDN line.

In order to avoid the entry of unwanted calls, the PortaNet terminal adapter allows the programming of call filters (BACKUP-TA_CNUM).

IX.10.1.2 PortaNet SLAVE operation

If the PortaNet works as SLAVE, the entry or exit to the backup mode will be guided by different criterions than the defined ones in the MASTER mode. First of all, it is necessary that the Backup option be enabled.

The unit will work by default in IP mode monitoring the state of the ISDN line continuously. If an incoming call is detected, the unit will switch to backup mode automatically. If the call hungs up, the unit will pass to work in IP mode again.

During the backup phase the PortaNet SLAVE goes on sending coded audio through the main line in order to the PortaNet MASTER can detect that the line has been recovered.

It is also possible to program call filters in the PortaNet SLAVE in order to avoid the unit answering unwanted calls.

PortaNet MASTER Backup Configuration

1.- Disable the backup operation: SYS - BACKUP – ENABLE - OFF

2.- Set the PortaNet as MASTER: SYS – MODE - MASTER

3.- Set the timers: SYS – TIMER – DOWN – x sec.

SYS – TIMER – UP – y sec.

4.- Set the backup audio mode and the ISDN numbers:

CONF – BOOK – INDEX 1 →

5.- Enable the Backp operation: BACKUP – ENABLE - ON

PortaNet SLAVE Backup Configuration

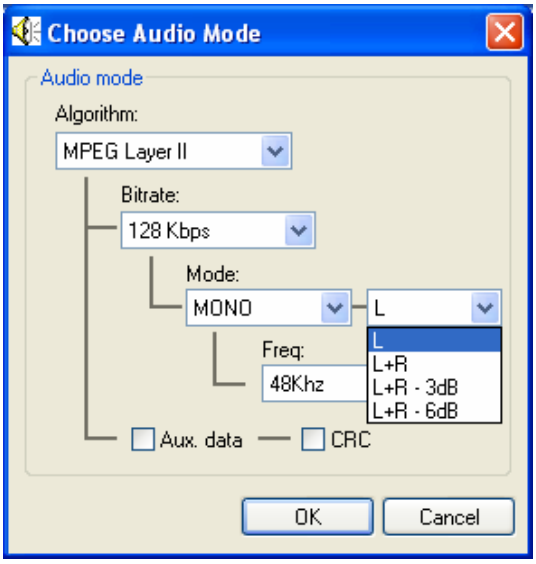
1.- Disable the backup operation: SYS - BACKUP – ENABLE - OFF

2.- Set the PortaNet as MASTER: SYS – MODE – SLAVE

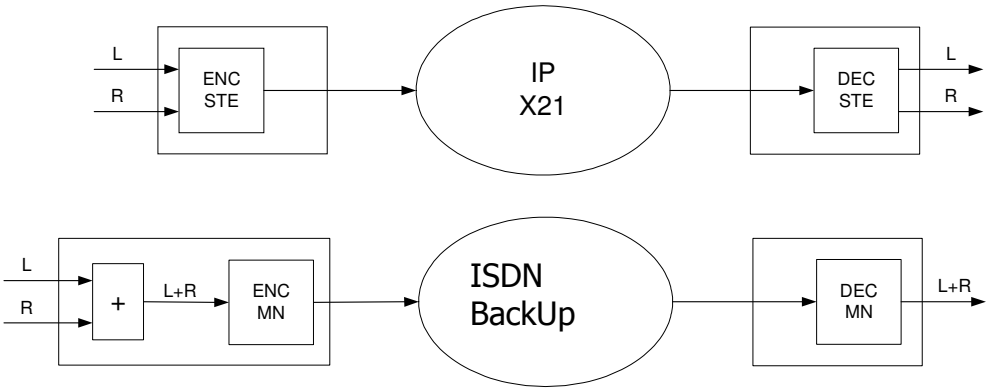
3.- Enable the Backp operation: BACKUP – ENABLE – ON

It is possible to select between the left channel L or L+R as audio source of the encoder 1, and R or L+R as audio source for encoder 2, when mono mode is selected for the backup (ISDN) connection.

This option is only available from the PortaNet web Page:



This option is very useful for an 64kbps ISDN link, where sending a stereo signal could involve a lack in the audio quality. Thus, encoding L+R signal, as a mono signal, will provide a much better audio quality. The diagram below explain this:



PortaNet Blocks

Knowing the different parts that comprise the PortaNet will help the user understand the various operational modes. The following chapter describes the different elements and their relationships.

These sections are arranged in two different groups: Communications and audio.

X.1 Communications

There are two different communication interfaces: ISDN and IP.

PortaNet can establish connections both via ISDN and IP networks. The procedure is very similar and can be carried out from the web page or the main menu. Basically, PortaNet operation modes are almost the same for both interfaces with the difference being the available bandwidth, which will place some limitations on the compression mode, and also on the encoder/decoder in line 2 over IP, which cannot be changed and will work in G722 mode.

There are two different independent communication channels

Both over ISDN and IP, it is possible to establish two independent communications. Thus, one of the connections will carry the program audio, and the second one will hold the coordination channel. The first line (line 1) is called program line, and the second one (line 2) TalkBack line:

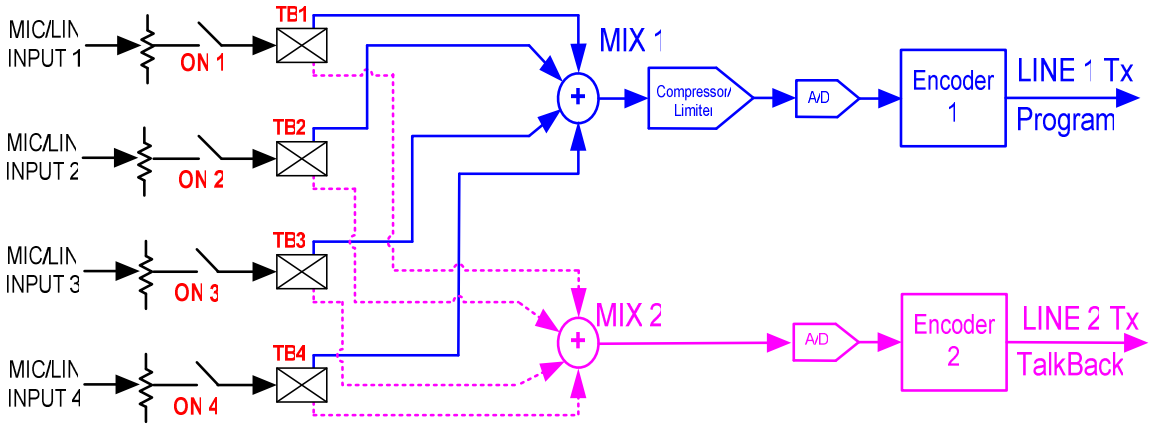
LINE 1 → PROGRAM CHANNEL:
Tx channel = **PROGRAM Tx**
Rx channel = **CUE**

LINE 2 → COORDINATION (TALKBACK):
Tx channel = **TALKBACK Tx**
Rx channel = **TALKBACK Rx**

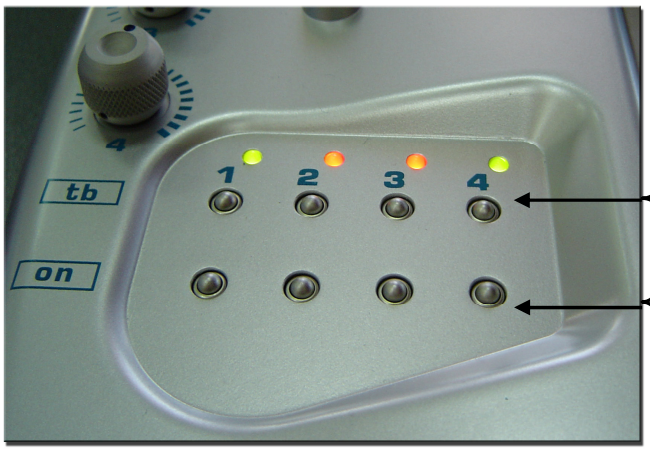
X.2 Audio MIC/LINE inputs

There are four MIC/LINE level audio inputs which can be assigned to either communication channel:

PortaNet is equipped with four MIC/LINE level audio inputs, and each input can be configured to accept LINE or MIC level signals. There are two mixers, one for each communication channel or line, and each audio input can be configured to be added to the audio contribution on either mix1 or mix2:



The TB key is used to determine which mixer the corresponding audio input will be added to.



These are the four TB keys corresponding to the four inputs of the unit.

The 'ON' key determines if the corresponding input will be enabled or disabled (muted).

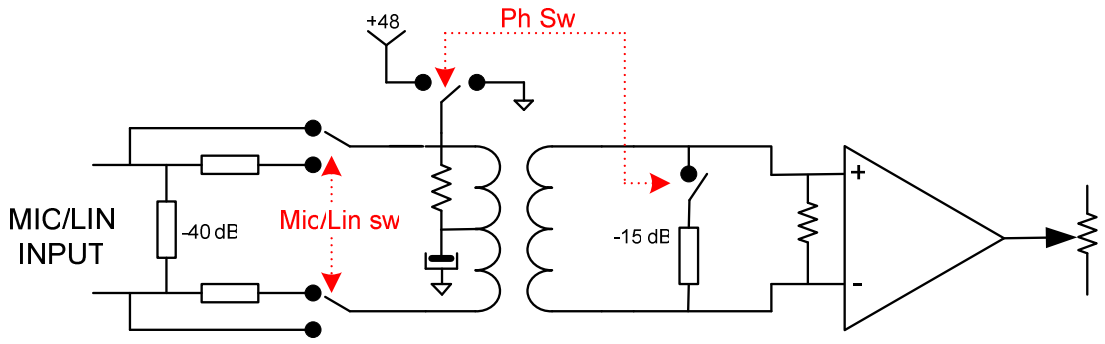
Differences between the audio on line 1 and line 2:

1. *The audio output of mix1 passes through a compressor/limiter before feeding line 1 encoder. Line 2 does not have a compressor/limiter.*
2. *The audio vu-meter ONLY monitors the output of MIX1.*
3. *In IP, the compression mode for line 2 is fixed to G722 mode.*

Audio input diagram

All the four inputs of the PortaNet are balanced XLR connector. Three of them are located on the front panel and the fourth one is on the rear panel. All of them can be configured to accept LINE or MIC levels, and to be phantom powered (48 volts.).

At either side of the connectors are microswitches which allow the selection between MIC/LINE levels and phantom power. On the top panel there is one level control for each input. When the phantom switch and the MIC level are selected, the audio input gain is decreased by 15 dB to compensate for the higher output of condenser microphones.



X.3 Audio "International Sound" input

This is an additional line level which can be mixed to the headphone outputs to provide the international sound to the commentators.

X.4 Audio outputs

PortaNet includes two kinds of audio outputs: headphone and balanced line level auxiliary outputs. These outputs can be configured depending on the requirements of the application, by setting the matrix available in the PortaNet web browser.

The audio signals on these outputs can be both the audio received and the audio contribution to either line.

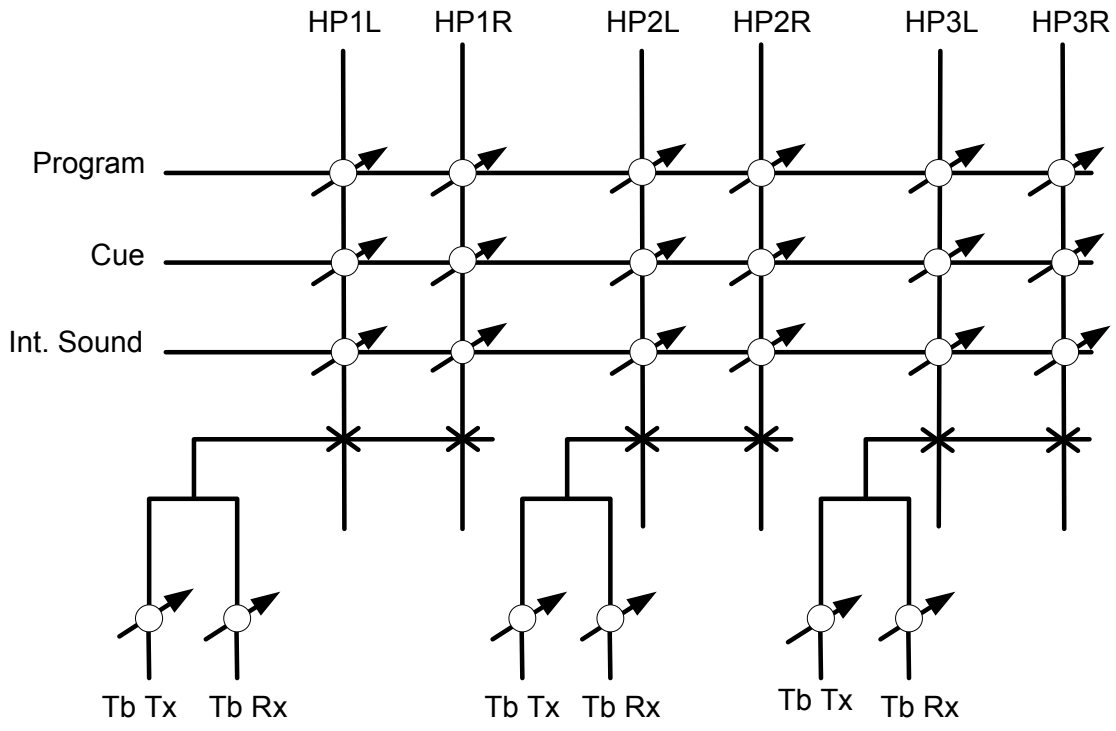
X.4.1 Headphone outputs

There are three headphone outputs, each one related to the corresponding MIC/LINE input. That means:

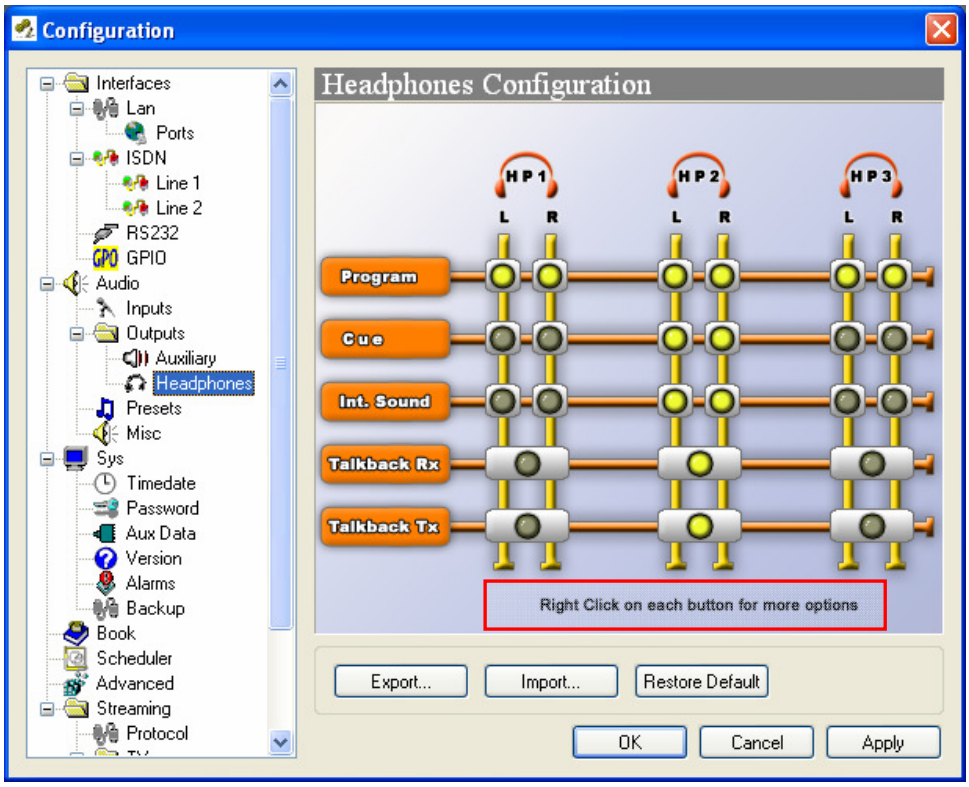
HP 1 ←→ MIC/LIN 1
HP 2 ←→ MIC/LIN 2
HP 3 ←→ MIC/LIN 3

This relationship means that the audio signals present on any headphone output will depend on the status of its corresponding input. In other words, it will depend on whether the input is connected to line 1 or line 2.

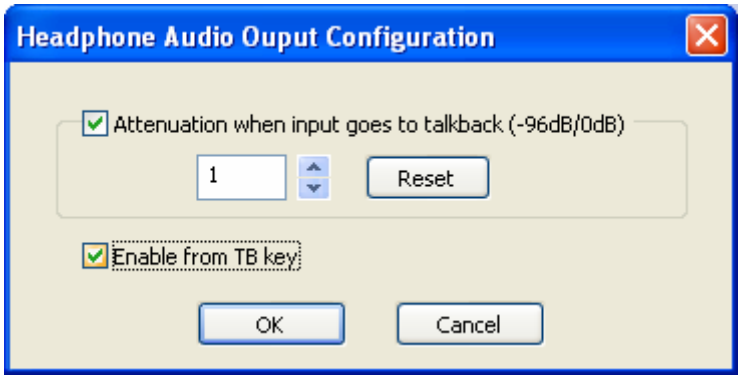
- The audio sources for each headphone can be selected from the PortaNet web browser. At the same time, it is possible to define whether some of them will be attenuated depending on which line (program or talkback) its corresponding input t is connected.
- The audio sources that can be selected are: Program, Cue, Talkback Tx, Talkback Rx and International Sound.
- Program, Cue and International Sound can be selected in either headphone ear independently.
- The following diagram shows how the headphone outputs can be configured:



Following is showed the dialog to select the headphones configuration.



By right clicking on each cross point a new dialog appears that allows the user to define the behaviour of the selected output depending on where the corresponding input is connected, that is, depending on if the MIC/LINE level input is connected to the program line or to the talckback line. Also, it allows the user to select whether the signal will be controlled from the audio input keypad.

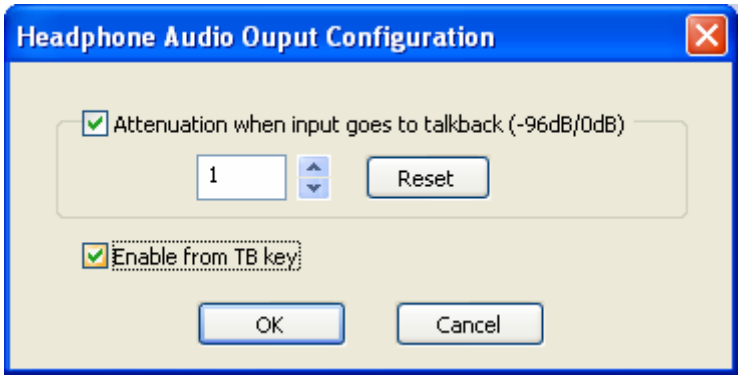


X.4.1.1 Attenuation when the MIC/LINE level input goes to Talkback or Program

It is possible to define an automatic attenuation on signals that feed the headphone outputs when the corresponding inputs go from Program to Talkback

or vice versa. The attenuation will be carried out according to the following critera:

Program, Cue or International Sound can be attenuated when the corresponding input is connected to the Talkback channel. It is possible to define an independent attenuation level for each of them.



The same thing can be done regarding Talkback Tx and Talkback Rx, that is, each signal can be attenuated when the corresponding input goes to program.

X.4.1.2 Enable from TB Key

"Enable from Tb Key" is an additional option that only is available for Talback Tx and Talkback Rx. When it is enabled, the corresponding signal will be sent to the headphone outputs only when the corresponding input has been connected to the Talkback channel, that is when its corresponding Tb key has been pressed.

When this option is disabled, this signal will be always available on the headphone outputs regardless of whether the corresponding input is connected to the Talkback channel or not.

X.4.2 Balanced audio outputs.

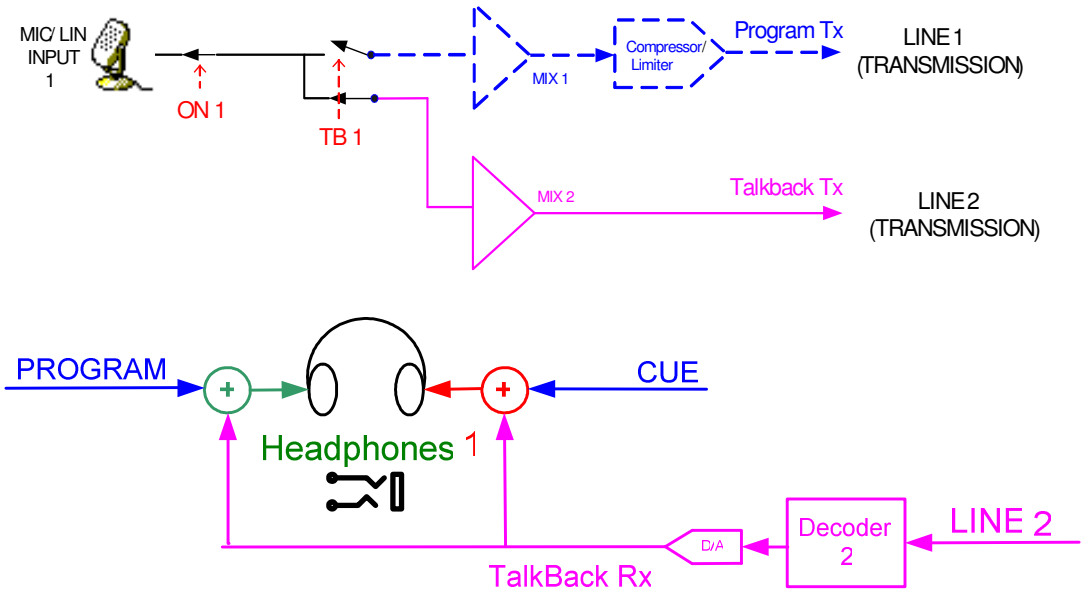
There are two line level auxiliary outputs. Both of them are balanced and the connectors are different due to space saving reasons. One of them, called 'AUX' is located on the front panel, and the other one, called 'REC' is on the rear one. **The different configurations and technical specifications are detailed in the chapter V - PortaNet audio outputs.**

Operation examples

XI.1 Using the TalkBack channel

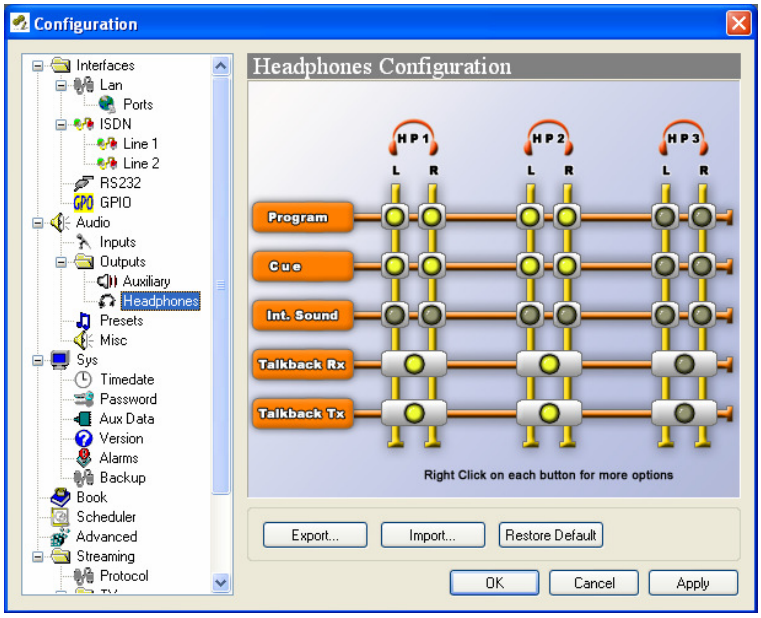
- **One of the commentators will connect to the studio using the TalkBack line:**

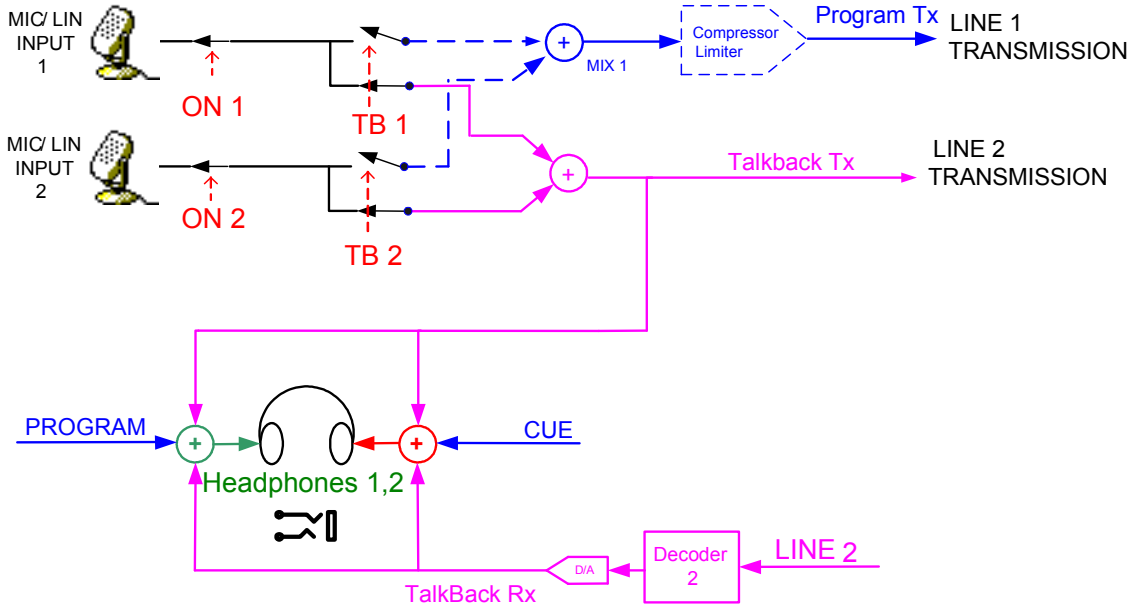
1. Make a call on line 2 (Call 2 key).
2. Press the TB key of the input where the talent's microphone is connected. The corresponding LED should be on RED.
3. The return TalkBack program will be received on both sides of the corresponding headphones if it was selected in the headphone configuration matrix together with Program and Cue signals.
4. The following diagram show the audio signals when an input is connected to the TalkBack program:



- **Several commentators are connected to the studio using the TalkBack line:**

1. Make a call on line 2 (Call 2 key).
2. Press the TB key of the input where the talent's microphone is connected. The corresponding LED should be on RED.
3. In order that the commentators are able to hear each other, Talkback Tx must be selected in the headphone configuration matrix. By selecting also the Talkback Rx signal, both the audio contribution and the received audio of the TalkBack line will be added on both ears, as it can be seen from the following diagram:

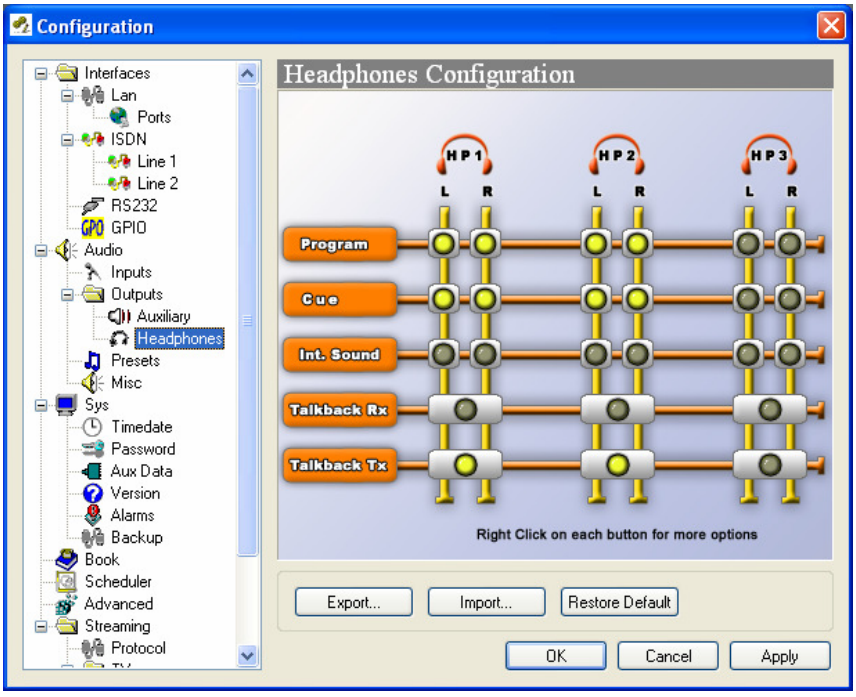


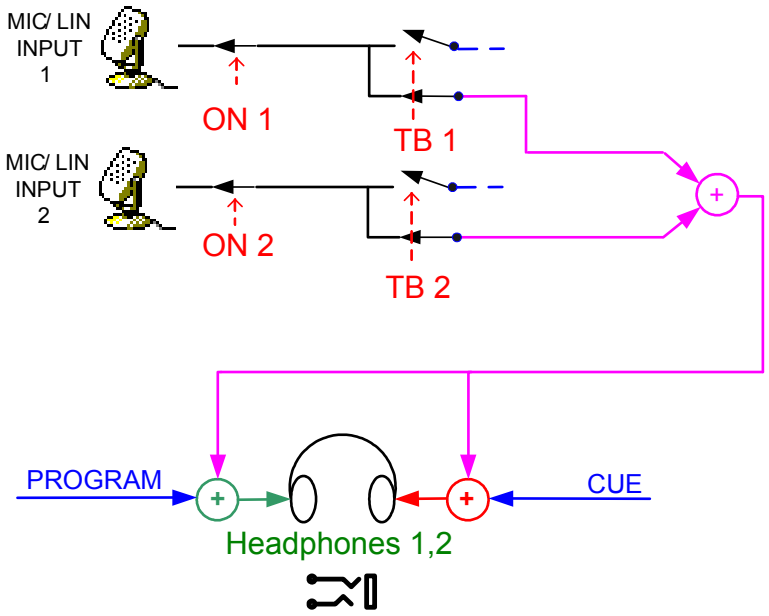


The fourth MIC/LINE input can use the 'aux' output as monitoring output.

- **The commentators want to talk to each other.**

1. Make a call on line 2 (Call 2 key).
2. Press the TB key of the inputs where the talent's microphones are connected. The corresponding LED should be on RED.
3. In order for the commentators to be able to hear each other, Talkback Tx must be selected in the headphone configuration matrix as it can be seen from the following diagram:



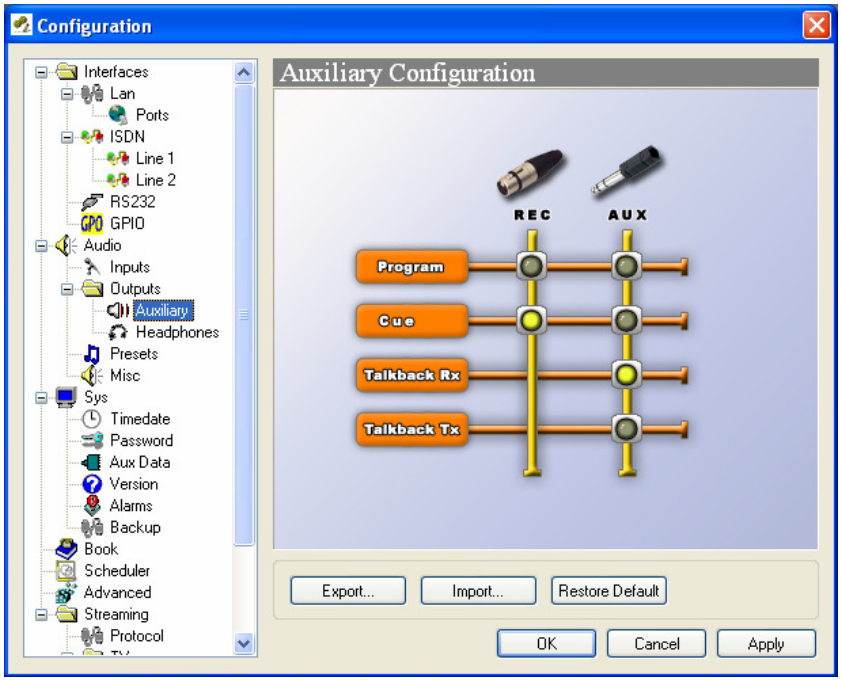


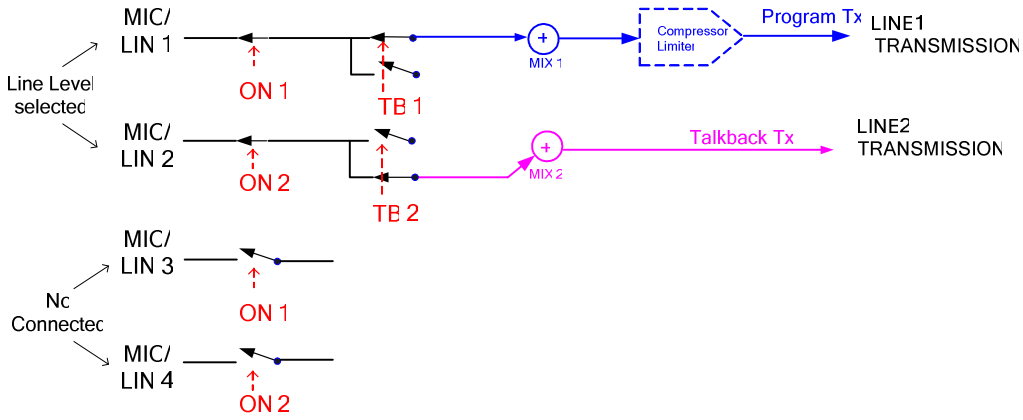
XI.2 Connecting to an external mixer

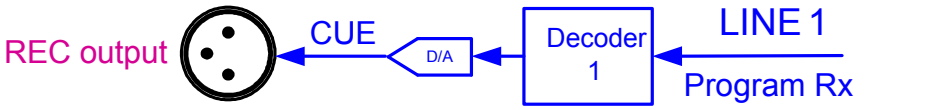
1. Configure two inputs to work with line levels.

2. One of the inputs must be connected to the TalkBack line by pressing the corresponding Tb key. This input will act as TalkBack audio channel. The other input will be connected to the program line (ON key) and will act as program audio channel.

3. The 'REC' and 'aux' outputs must be configured from the Auxilliary Outputs menu in the following way:



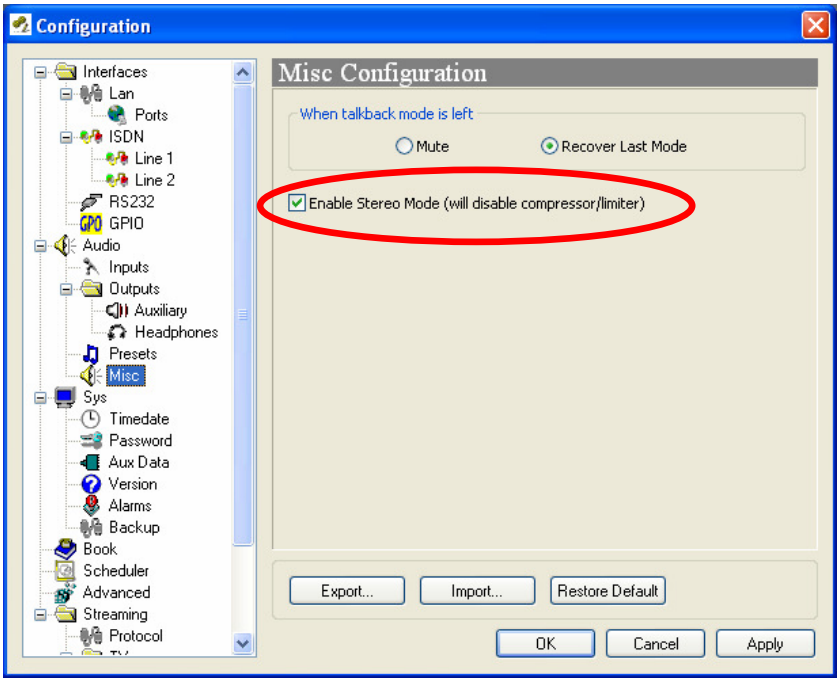




XI.3 Stereo operation mode:

This is a special mode where the compressor/limiter is disabled. The following steps must be followed to work in stereo mode:

1. Go to the "Misc" option in the Audio configuration menu and enable the Stereo mode. After that, the compressor/limiter will be disabled:

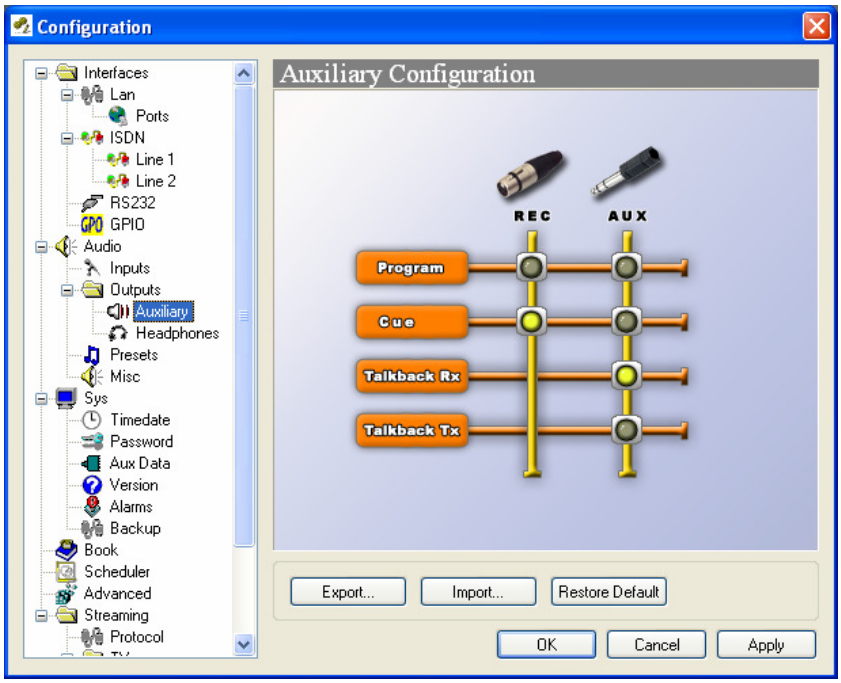


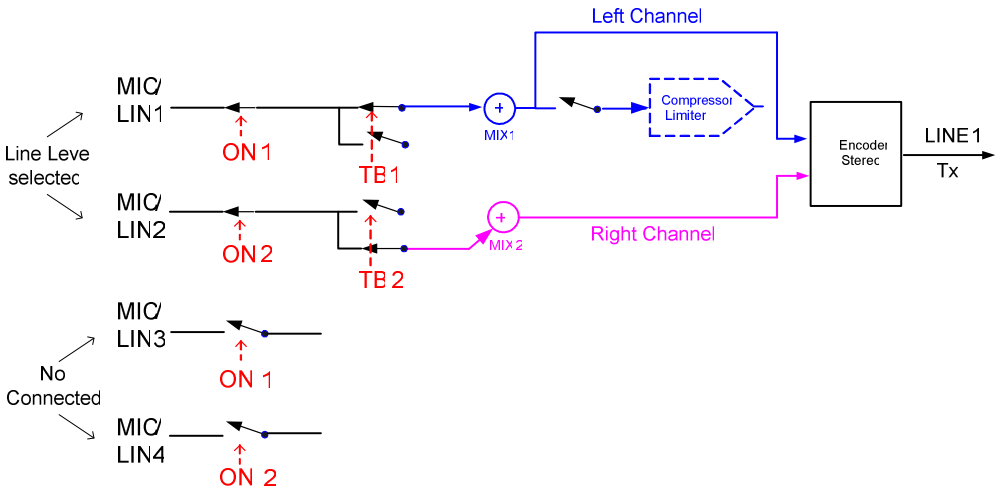
2. Configure two inputs to work with line levels.
3. Disable those inputs which are not going to be used.
4. The left channel audio source will be the audio input assigned to the program line (led on green).
5. The right channel audio source will be the audio input assigned to the TalkBack line(led on red).

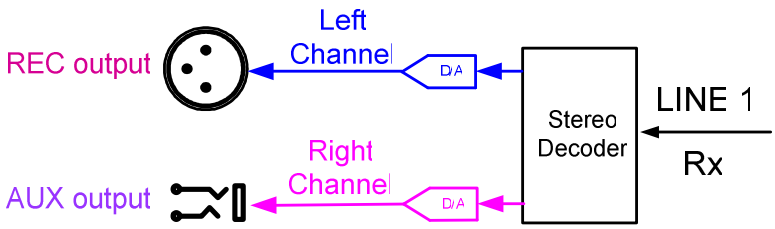
The audio source for the left and right channels can come from several inputs: when several line level inputs are assigned to the same line, they will be added before contributing to the corresponding channel.

When in stereo mode and using IP it is not possible to use line 2. However, over ISDN it is possible to make 128Kbps stereo communications, i.e. 2x64Kbs channels, so this restriction does not apply to ISDN.

6. The left audio channel of the received audio will be present on the 'REC' output, whilst the right audio channel will be present on the 'AUX' channel. The output matrix must be configured as follows:


Configuration
Interfaces
Lan
Ports
ISDN
Line 1
Line 2
RS232
GPIO
Audio
Inputs
Outputs
Auxiliary
Headphones
Presets
Misc
Sys
Timedate
Password
Aux Data
Version
Alarms
Backup
Book
Scheduler
Advanced
Streaming
Protocol
Auxiliary Configuration
REC
AUX
Program
Cue
Talkback Rx
Talkback Tx
Export...
Import...
Restore Default
OK
Cancel
Apply


Left Channel
MIC/
LIN1
Line Leve
selectec
ON 1
TB 1
MIX1
Compressor
Limiter
Encoder
Sterec
LINE 1
Tx
MIC/
LIN2
ON 2
TB 2
MIX 2
Right Channel
MIC/
LIN3
No
Connected
ON 1
MIC/
LIN4
ON 2


REC output
Left
Channel
D/A
Stereo
Decoder
LINE 1
Rx
Right
Channel
AUX output
D/A

Problem-solving guide

This chapter is aimed at providing some solutions to common problems when using Prodys Codecs.

XII.1 Audio problems

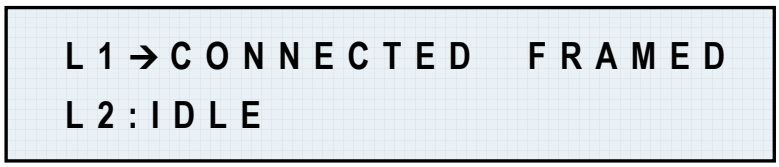
Audio problems and some possible solutions:

XII.1.1 No Audio on the outputs

When audio is not present on any output and before proceeding, it is convenient to check the audio matrix configuration in order to make sure that the audio sources has been correctly selected.

XII.1.2 The program line is connected but there is no audio on the outputs

In order that audio outputs carry audio, the corresponding decoder must be synchronized, that means, the decoder has to detect the format of the bit stream from the audio data being received. This auto-detection is automatic, and this situation is represented in the screen with the word "FRAMED".



Thus, when the decoder is not 'framed', there is no audio on the outputs.

Some things that can cause the decoder not to be synchronized:

1. The communication line is not working properly.
2. The encoder at the other end is not configured in a compatible mode, or it is not encoding properly.
3. The equipment may be faulty. The easiest way to test it is to make a 'loop call', by calling from the unit to itself and check if everything works correctly.

If the decoder is 'framed' but there is still no audio on the outputs:

1. There is no audio present on the inputs of the codec at the other end, or the audio in the other codec is not well configured.
2. The codec at the other end is not working properly.
3. The equipment may be faulty. The easiest way to test it is to make a 'loop call', by calling from the unit to itself and check if everything work fine.

XII.1.3 There is no audio output at either end

Some things that can cause these problems are the following:

1. The communication line is failing.
2. The remote codec is not working properly.
3. The equipment may be faulty. The easiest way to test it is to make a 'loop call', by calling from the unit to itself and check if everything work fine.
4. The codecs involved in the communication are not compatible.
5. Both codecs are configured in AUTO mode (automatic encoder mode).
6. The audio settings are wrong configured in both codecs.

XII.1.4 Audio distortion

It is possible to modify the gains on inputs and outputs, so check if this has been set properly. The audio input and output levels can be monitored from the display by pressing the INF key, or from the web page. Also, the input level for the program line (line 1) is indicated on the vu-meter on the top panel.

Another cause could be that the other end is using digital audio such us AES/EBU as an audio source, and no AES/EBU signal is being sent, or the equipment is faulty.

XII.2 ISDN communication problems

Here follows some common problems and solutions when communicating via ISDN networks. Applicable to some Prodys Codecs.

XII.2.1 Unit cannot make outgoing calls

1. If the equipment is not connected correctly to an ISDN socket, it cannot make or receive any calls. Even though the ISDN connection is apparently correct, the line status can be checked by making a call. If the line does not respond properly, the message "NO PHYSICAL LINE" is displayed.

   Some ISDN lines work in a way such that there is no TEI (Terminal End-point Identifier) assigned until a first call is attempted.

2. The ISDN protocol does not correspond to that of the provided line.

XII.2.2 Unit cannot receive incoming calls

1. If the equipment is not connected correctly to an ISDN socket, it cannot make or receive any calls. Even though the ISDN connection is apparently correct, the line status can be checked by making a call. If the line does not respond correctly, the message "NO PHYSICAL LINE" is displayed.

2. If the equipment is configured with a local number (LNUM) for a line, it will reject any call to any other number without notifying the user. The local number serves to differentiate between multiple devices connected to the same number. The solution is to select the NONE option in the terminal adapter configuration menu.

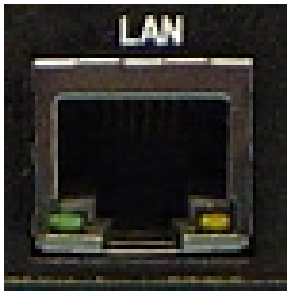
3. If the equipment's call filters are activated for a line (CNUM activated), all calls to that line will be rejected if they do not match those of the filter.

4. If the ISDN network requires the configuration of the SPID numbers, these have to be configured according to the provider instructions for the unit to work properly over the link. These numbers can be configured both from the control keypad and from the web page.

XII.3 IP communication problems

XII.3.1 Prodys Codec's Web Page cannot be accessed

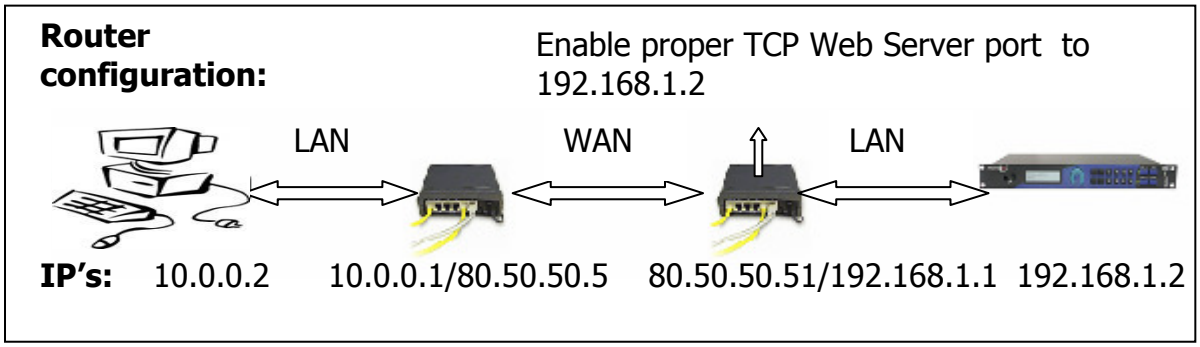
1. The cable or the connector are faulty. There are some leds on the rear panel to check the Ethernet link:



- Green LED → LINK STATUS: ON = Connected
- Orange LED → RECEIVE STATUS: On = Receiving Data.

2. The IP address of Prodys Codec and the PC from which we are trying to access the web page are not in the same subnet, even although they are connected to the same LAN. The default factory settings for the IP address and netmask are 192.168.100.100 and 255.255.255.0 respectively. The user must change the IP settings in the computer or in Prodys Codec to match the same network.
   To change the IP address of Prodys Codec, the user can use the control keypad or the web page. In the menu, it is set by selecting CONF-PORTS-LAN. The IP settings can be entered manually or they can be obtained automatically when the unit starts from a DHCP server.
   Once the IP setting on the PC and the Prodys Codec are configured properly, you can check that IP connectivity exists by typing the following command at the command prompt of the operating system: C:>ping 192.168.100.100 ↵. This tool will inform the user whether there is IP connectivity between the PC and Prodys Codec or not. If there is IP connectivity bur the problem still persists please refer to point 4.

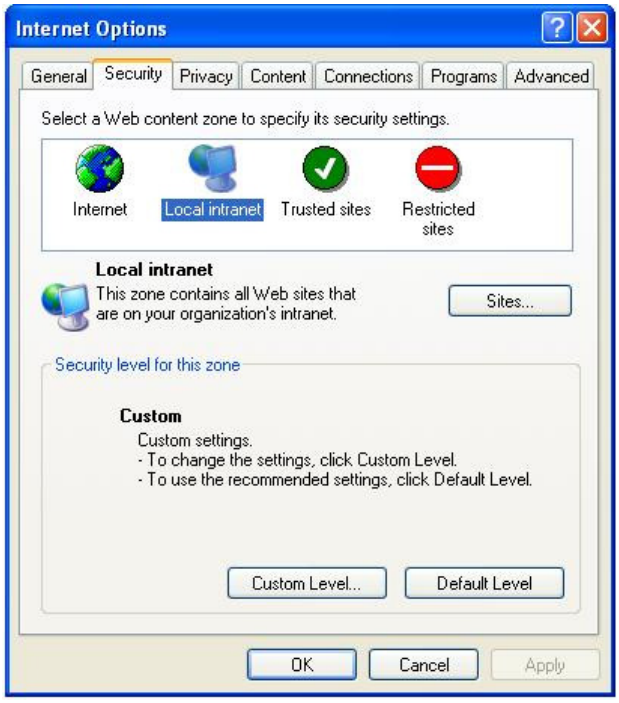
3. The PC and Prodys Codec are not connected to the same LAN, but connected through a router. This connection could be, for example, a connection over the Internet with DSL routers. In this case, to access and operate the Prodys Codec, the router should open the a "Web Server Port" (this is by default HTTP port 80) and a range of 30 TCP ports from the "Base Port" on (default from 50011 until 50041) and forward this traffic to the Prodys Codec IP address. See pop up window of section "VIII.2.1.1.1-PORTS" for setting these equipment parameters.



4. Microsoft Internet Explorer blocks ocx installation when trying to access Prodys Codec web page from microsoft internet explorer. The first time the user accesses the Prodys Codec web page, an OCX file has to be downloaded and installed in the computer. This is done automatically unless the web browser disables it. So, depending on the configuration of the web browser, the following message can appear when first accessing the Prodys Codec web page:

Internet Explorer - Security Warning
Windows has blocked this software because it can't verify the publisher.
Name: SClusterOutputBox.cab
Publisher: Unknown Publisher
OK
To help protect your computer, Windows will block software when it cannot verify the publisher. See how this helps protect your computer.

Go to Internet Options in IExplorer, click on 'Security' tab, and set 'prompt' when downloading ActiveX signed and unsigned controls at Local and Internet zones.

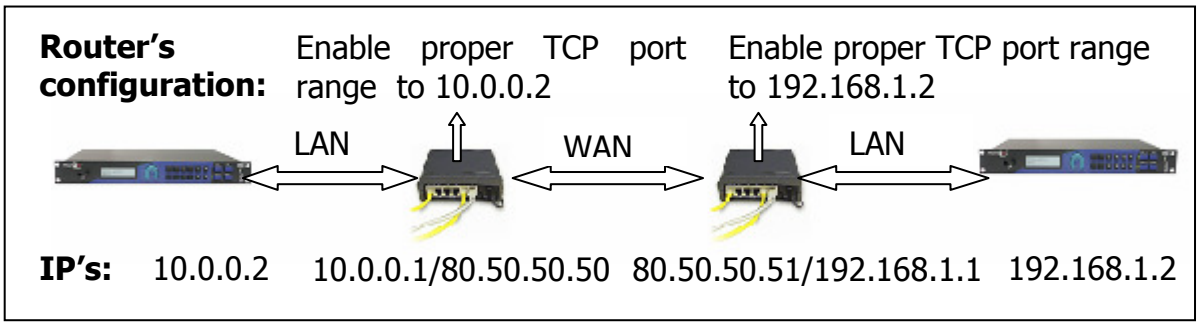


Each firmware version might have a different OCX file, so the new OCX should be installed as it is done for the first access to the web page of the unit. If the unit was upgraded and, depending on the 'cache' configuration of the Internet explorer, there might be problems when accessing the web page, given that the old web page might be offered by the browser instead of the real one, which should be installed to replace the old one. In this case, a message indicating 'Incorrect Versions' will appear as soon as the user click on 'Control' or 'Monitor' on the login page. Click on F5 to skip the cache entries, and access to the 'real' web page. Even after pressing F5 and, depending on the IExplrorer configuration and/or version, this situation might continue. In that case, go to Internet Options in IExplorer, click on 'General' tab, and delete temporary files.

Windows Vista: Should the user experience a problem when downloading the OCX file when first accessing the web page of the unit, please disable UAC (User Access Control) on Windows Vista. Once the OCX file has been installed in the computer, UAC can be enabled again.

XII.3.2 When connecting two audiocodecs in unicast, there is no audio at one end.

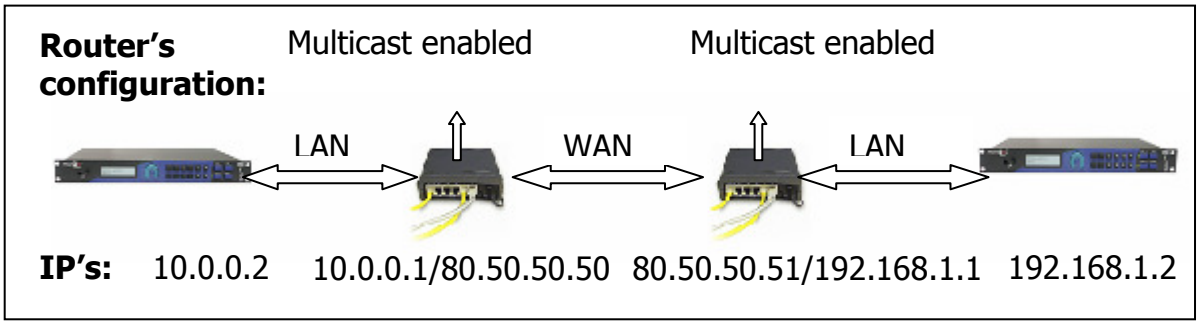
Please check the enabling of the full TCP port range ("Base port" + following 30 ports) of any router or firewall between the audiocodecs. Hint: start checking from the disturbed audiocodec on.



XII.3.3 No audio when connecting two audiocodecs using Multicast

1. Please check the enabling of the full TCP port range ("Base port" + following 30 ports) of any router or firewall between the audiocodecs.

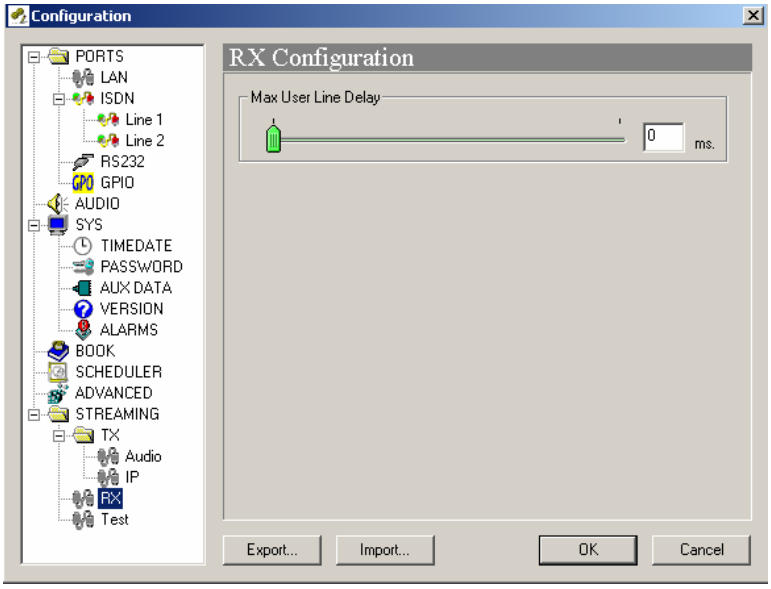
   Take into account that only a small part of the Internet called Mbone supports multicast traffic, so to send multicast traffic over the Internet, a technique called 'tunneling' must be used. VPN networks can be used for this purpose.



XII.3.4 Interruptions to audio when connecting two Prodys Codecs.

1. A decisive factor in real time audio streaming is the 'jitter', or delay variation. To deal with the jitter in the connection, PRODYS provides a tool which allows the user to modify the size of the reception buffer, and so, to compensate for the jitter. The maximum value for this buffer is 10 sec. This buffer has to be configured from the web page to, at least, the same value as the 'jitter' in milliseconds.

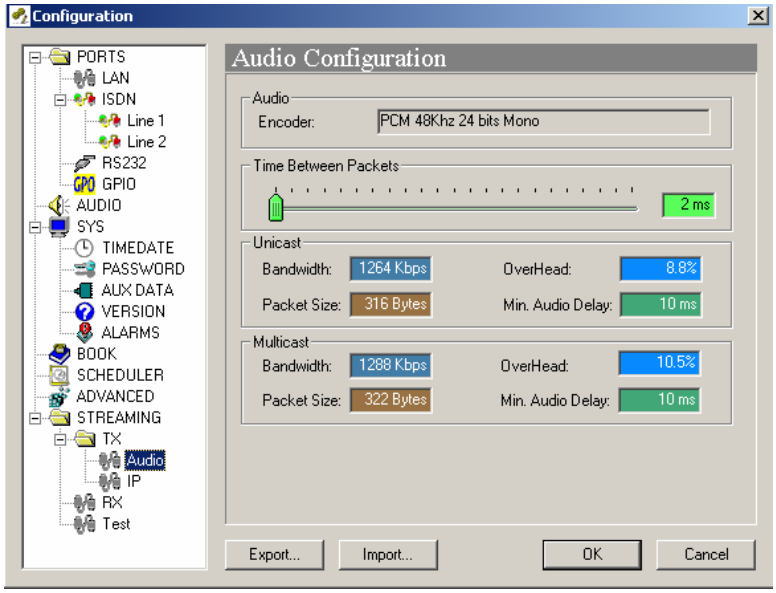
   The 'jitter' can be measured from the 'Test streaming' tool provided from the web page.



Note: With the Real Time Network Analyzer, the user can get information about the jitter during the audio connection in real time. For more information about this, please refer to chapter IX.6.7 – Proprietary protocols v2.

2. The audio interruptions could be due to a reduction on the bandwidth in the IP connection. With the 'Test Streaming' tool from the web page, the user can measure the download and upload bandwidth between two Prodys IP Codecs. Once the user knows the available bandwidth, it is possible to select the proper bit rate for the compression mode.

   The user can obtain information about the delay of the encoder/decoder process for any particular mode, and the actual bandwidth which will be required for that mode, from the web page, in the 'audio' tab. Besides, in the non-block modes, like PCM or apt-X, it is possible to modify the size of the frames from 2 to 24 msc. In the rest of the modes, this size is fixed and determined by the corresponding standard. The larger the block size, the higher the delay, but the smaller the required bandwidth (more efficient use of IP packets), and vice versa.



3. Lost and/or disordered packets might be the cause of audio interruptions. To obtain information about these parameters in real time, the Real Time Network Analyzers can be used. For more information about this, please refer to chapter IX.6.7 – Proprietary protocols v2.

Technical specifications

Audio Interfaces

4 Mic/line inputs:

- XLR connectors.
- Maximum levels:
    - LINE levels: +20dBu.
    - MIC levels: -46dBu (without compressor).
    - MIC levels: -33dBu (with compressor).
- Input impedance:
    - LINE mode = 40Kohm.
    - MIC mode = 40Kohm.
- Mic input sensitivity adjustable from -25 to -65 dBu.
- THD+N < 0.015% (MIC mode), 0.01% (LINE mode). @ 1kHz.
- SNR > 80dB (MIC mode), 95dB (LINE mode).
- Phantom power (48 Volts).

International Sound input:

- XLR connector.
- Maximum level: +20dBu.
- Input impedance: 40Kohm.
- THD+N < 0.01% @ 1kHz.
- SNR < 80dB 95dB .

3 headphone outputs:

- 6.3mm Jack stereo connectors.
- Bandwidth 20Hz-20kHz (1dB).
- Maximum output level: 6dBu (8 mW) over 300 ohm.
- THD+N < 0.01% @ 1kHz.
- SNR < 95 dB.

LINE level outputs:

- XLR male connector (REC output) and 6.3mm Jack stereo connector (AUX output).
- Bandwidth 20Hz-20kHz (0.3dB).
- Maximum output level: +20dBu.
- THD+N < 0.005% @ 1kHz.
- SNR < 100 dB.

Audio Compression

- G722.
- G711 A/μ Law.
- MPEG 1,2 layer II (ISO/IEC 11172-3 /13818-3).
- MPEG 1,2 Layer III (ISO/IEC 11172-3 /13818-3).
- MPEG 2 AAC LC (ISO/IEC 13818-7).
- MPEG 4 AAC LC (ISO/IEC 14496-3).
- MPEG 4 AAC LD & HE (ISO/IEC 14496-3).
- Standard and Enhanced aptX™.

BANDWIDTH (KHz)

Legend:
x = Not available in the standard.
#,*,- = Not implemented

- **MPEG 1,2 LAYER II (ISO/IEC 11172-3 /13818-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
32	4	X	X	4,9	x	x	7,3	#	#
64	10,7	4	4,8	11,7	4,9	6,1	11,3	7,3	11,3
128	20	10,7	16,3	15	11,7	13,6	11,3	11,3	11,3
192	20	14,5	20	15	15	15	x	x	x
256	x	20	20	x	15	15	x	x	x
384	x	20	20	x	15	15	x	x	x

Bit Rate	Fs=16 KHz		
	Mono	Stereo /Dual	JStereo
32	7,5	#	#
64	7,5	7,5	7,5
128	7,5	7,5	7,5
192	x	x	X
256	x	x	X
384	x	x	X

- **MPEG 1,2 LAYER III (ISO/IEC 11172-3 /13818-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
32	8,1	#	#	8,2	#	#	8,1	#	#
64	15,2	8,1	8,1	15	8,2	8,2	11,3	8,1	8,1
128	18,2	18,2	18,2	15	15	15	11,3	11,3	11,3
192	20	20	20	15	15	15	x	x	x
256	20	20	20	15	15	15	x	x	x

Bit Rate	Fs=16 KHz		
	Mono	Stereo/Dual	JStereo
32	7,5	#	#
64	7,5	7,5	7,5
128	7,5	7,5	7,5
192	x	x	X
256	x	x	X
384	x	x	X

- **MPEG 2 AAC LC (ISO/IEC 13818-7)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	15,8	7,5	8,3	14	8	12	11,3	8,3	10,5
128	20	15,8	15,8	15	14	15	11,3	11,3	11,3
192	20	15,8	15,8	15	14	15	x	11,3	11,3
256	20	20	20	x	15	15	x	11,3	11,3
384	x	20	20	x	15	15	x	x	x

- **MPEG 4 AAC LC (ISO/IEC 14496-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	18	7,5	8,3	15	8	12	11,3	8,3	10,5
128	20	18	20	15	15	15	11,3	11,3	11,3
192	20	18	20	15	15	15	x	11,3	11,3
256	20	20	20	x	15	15	x	11,3	11,3
384	x	20	20	x	15	15	x	x	x

Bit Rate	Fs=16 KHz		
	Mono	Stereo/Dual	JStereo
64	7,5	7	7
128	x	7,5	7,5
192	x	x	X
256	x	x	X
384	x	x	X

- **MPEG 4 AAC LD (ISO/IEC 14496-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	10,6	#	#	11,8	#	#	11,3	#	#
128	18,4	10,6	13,6	15	15	15	11,3	11,3	11,3
192	18,4	10,6	13,6	15	15	15	11,3	11,3	11,3
256	20	18,4	18,4	15	15	15	11,3	11,3	11,3
384	20	18,4	18,4	15	15	15	-	-	-

- **MPEG 4 AAC HE (ISO/IEC 14496-3)**

Bit Rate	Fs=24KHz			Fs = 16 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
24	12	#*	#	12,3	#	#
32	16,8	#	#	15	#	#
48	16,8	12	15	15	12,3	15
56	16,8	12	15	15	12,3	15
64	20	16,8	20	15	15	15
128	20	20	20	x	15	15

- **aptX™ SD & ENH 16 bits**

Bit Rate	Fs=48KHz		Fs = 32 KHz		Fs=16 KHz	
	MN	ST/Dual	MN	ST/Dual	MN	ST/Dual
64	x	x	X	x	7.5	X
128	x	x	15	7.5	X	X
192	20	x	X	x	X	X
256	x	x	X	15	X	X
384	x	20	X	x	X	X

- **aptX ENH 20 bits**

Bit Rate	Fs=48KHz		Fs = 32 KHz	
	MN	ST/Dual	MN	ST/Dual
160	x	x	15	x
320	x	x	X	15
240	20	x	X	X
480	x	20	X	X

- **aptX ENH 24 bits**

Bit Rate	Fs=48KHz		Fs = 32 KHz	
	MN	ST/Dual	MN	ST/Dual
192	x	x	15	x
384	x	x	X	15
288	20	x	X	X
576	x	20	X	X

- **G722**

Bit Rate	Fs=16KHz
	MN
64	7.6

- **G711**

Bit Rate	Fs=8KHz
	MN
64	3.8

IP Protocols and compatibility

- DNS
- HTTP
- ICMP
- IGMPv2
- IPv4 /TCP /UDP :RFC 791, RFC 793, RFC 768, RFC 1112
- RIPv2
- RTP
    - :RFC 3550, RFC 3551, RFC 2250, RFC 3119,
    - :RFC 3190, RFC 4184, RFC 3555, RFC 3640
- SAPv1 :RFC 2974
- SDP :RFC 4566, RFC 3264 (EBU Tech 3326 Standard)
- SIPv2 :RFC 3261
- SNMPv2
- SNTP

Comunication ports

ISDN

- Protocols: EISDN, AT5ESS, DMS100 and NAT.
- 1 BRI connection. S/T and U interfaces.
- **BackUp system:** ISDN as a BackUp for IP or X21 links.
- RJ45 connector.

LAN port

- 10/100Base-TX Ethernet with auto-negotiation.
- RJ-45 connector.

GPIO port

- DB9F connector.
- 2 ground contact inputs.
- 2 NA/NC relay outputs.
    -Max voltage: 125VAC (0.5A), 110VDC (0.3A), 30VDC (1A).
    -Max. current: 2A.

RS232 port

- Sub-D 9 ways Female connector.

Power supply.

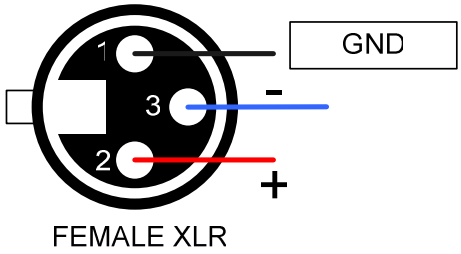
- Universal Input Range.
- AC input range: 100-240 V.
- Frequency range: 50-60 Hz.
- Output Power: 30W (with battery in charge status).
- Optional battery.

Weight and dimensions

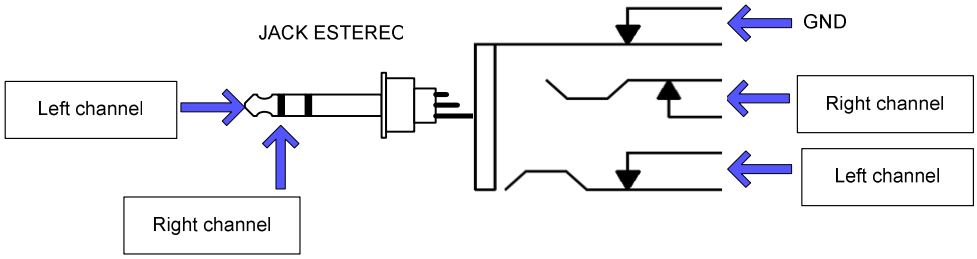
- Height: 70 mm; Width: 285 mm.; Depth: 243 mm.
- Weight: 2 Kg.

Connectors

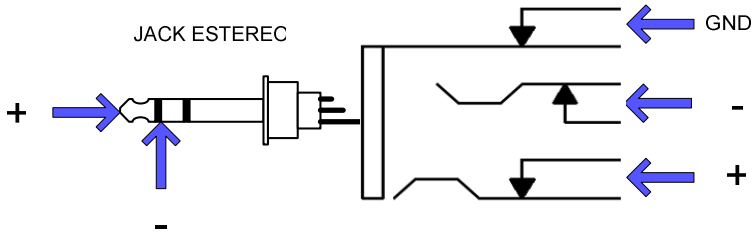
XIV.1 Audio inputs



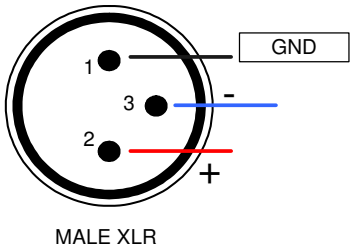
XIV.2 Headphone outputs



XIV.3 Aux output



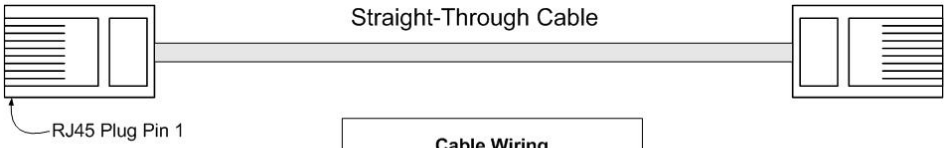
XIV.4 Rec output



XIV.5 LAN port

XIV.5.1 Connecting to a HUB or SWITCH

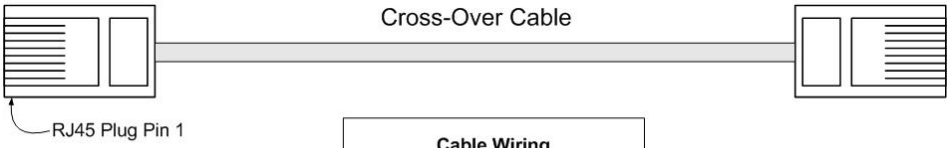
In the majority of cases you can simply connect the unit's LAN port to your Ethernet network's Hub or Switch using an Ethernet cable (CAT5). In this case you should use a standard 'straight-through' Ethernet cable (not a 'cross-over' cable). This kind of cable can normally be found in any IT shop. In any case, this cable is described in more detail below:



Cable Wiring	
Prontonet	**Hub**
Tx+ 1	1 Tx+
Tx- 2	2 Tx-
Rx+ 3	3 Rx+
Rx- 6	6 Rx-

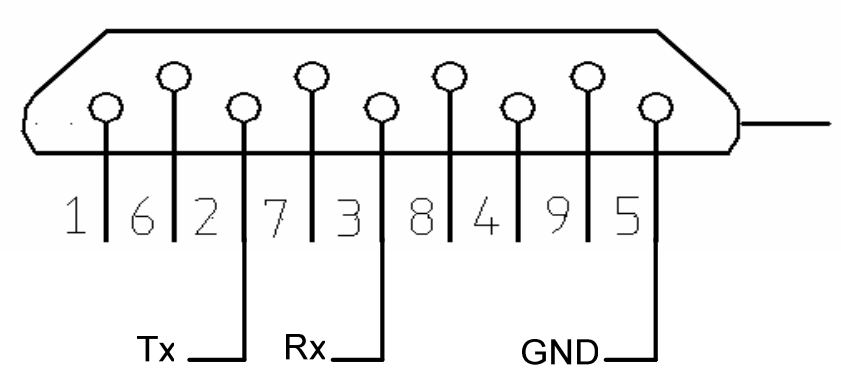
XIV.5.2 Connecting to a PC

In some cases, such as when you configure the equipment, it is possible that you will want to connect the unit directly to a PC. In this case the PC must have a free Ethernet port to connect to and you must use a 'cross-over' Ethernet cable. Again, any good IT shop will stock these cables. This time the wiring is as follows:



Cable Wiring	
Prontonet	**PC LAN card**
Tx+ 1	3 Rx+
Tx- 2	6 Rx-
Rx+ 3	1 Tx+
Rx- 6	2 Tx-

XIV.6 RS 232 port



The PortaNet acts as a DCE device, therefore the connection to each of the RS232 ports is wired in the following way:

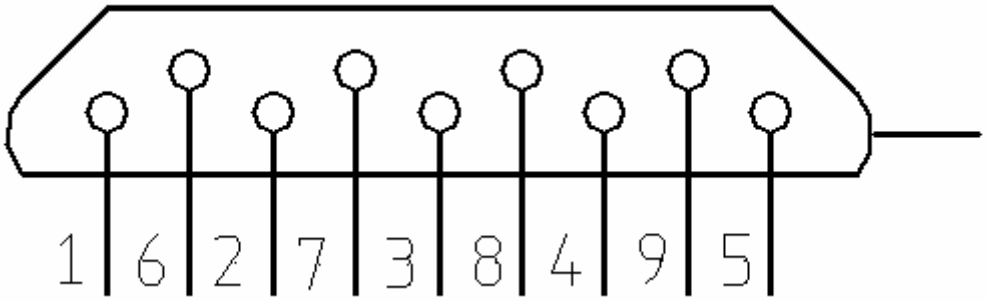
PortaNet – Pin 2.....Pin 2 PC

PortaNet – Pin 3.....Pin 3 PC

PortaNet – Pin 5.....Pin 5 PC

The PortaNet ignores hardware handshaking signals.

XIV.7 GPIO Port



Pin 1	Input 1		
		Pin 6	Input 2
Pin 2	GND		
		Pin 7	Normally Closed Relay 1
Pin 3	Common Relay 1		
		Pin 8	Normally Open Relay 1
Pin 4	Normally Closed Relay 2		
		Pin 9	Common Relay 2
Pin 5	Normally Open Relay 2		

Updating the firmware

It is posible to update several units at once with the HeraFlashPlus utility. The unit configuration won't be lost when updating.

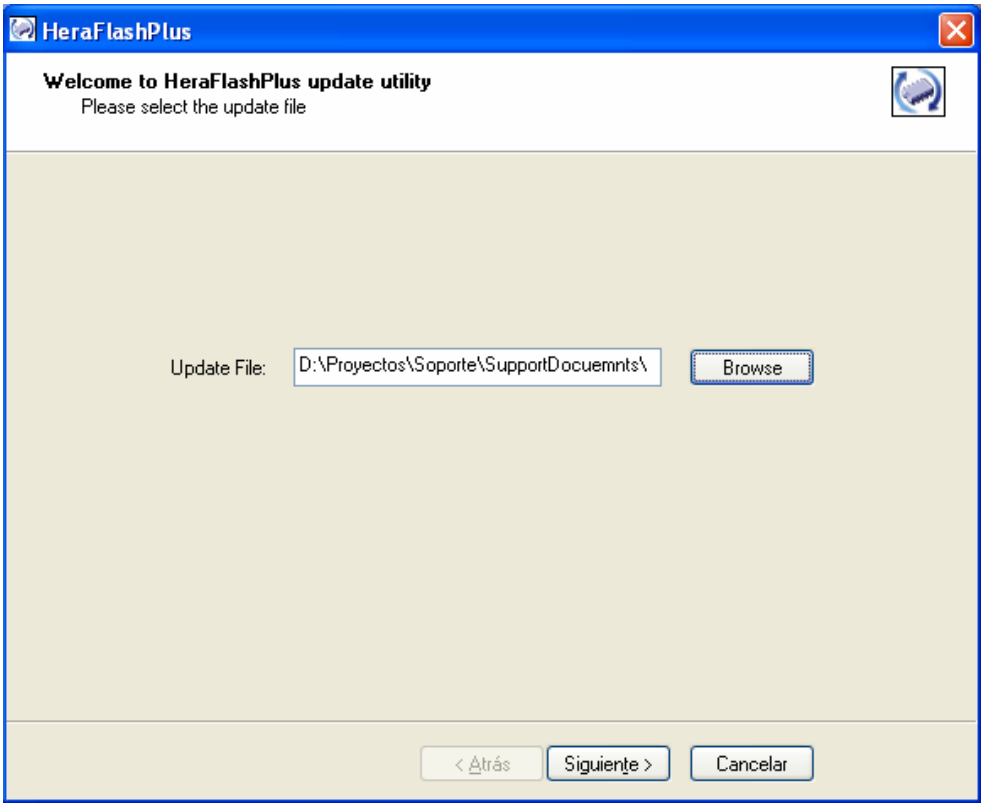
From version 5.2.1 on and using HeraFlashPlus version 2.5.0 or higher, the upgrading can be performed over WAN networks, such as Internet.

From version 5.4.1 onwards, all Prodys IP codecs will share the same firmware file, making it more easier to upgrade several different units to the latest firmware version.

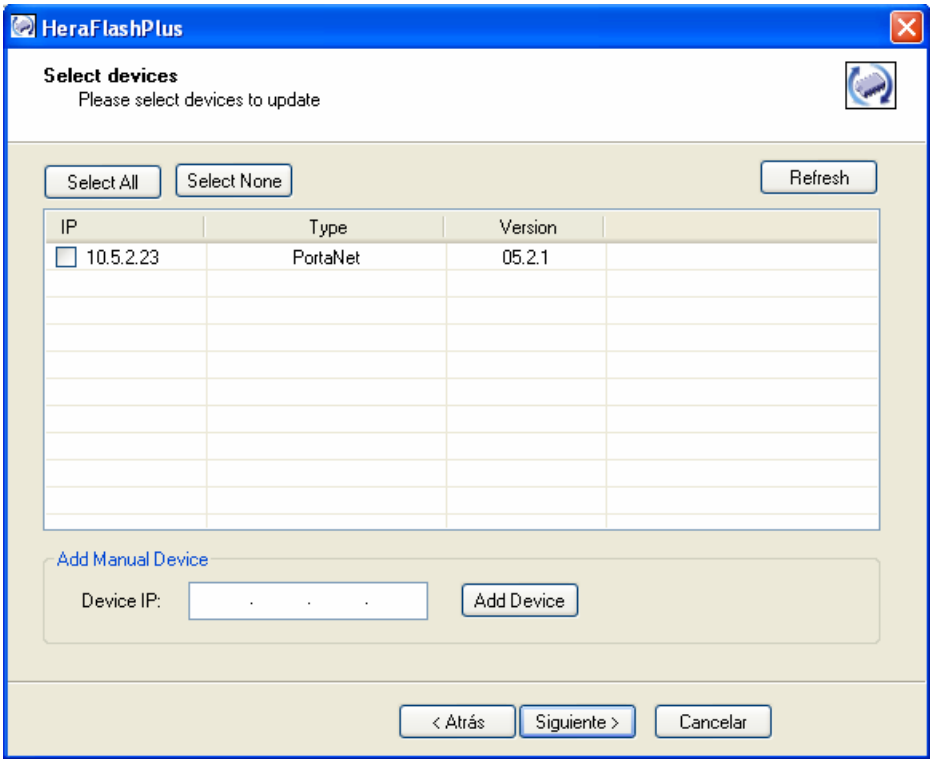
To update the firmware of a defined unit follow these steps:

1. Start up the HERAFlashPlus application.

2. Click on browse and select the file with the new firmware version.



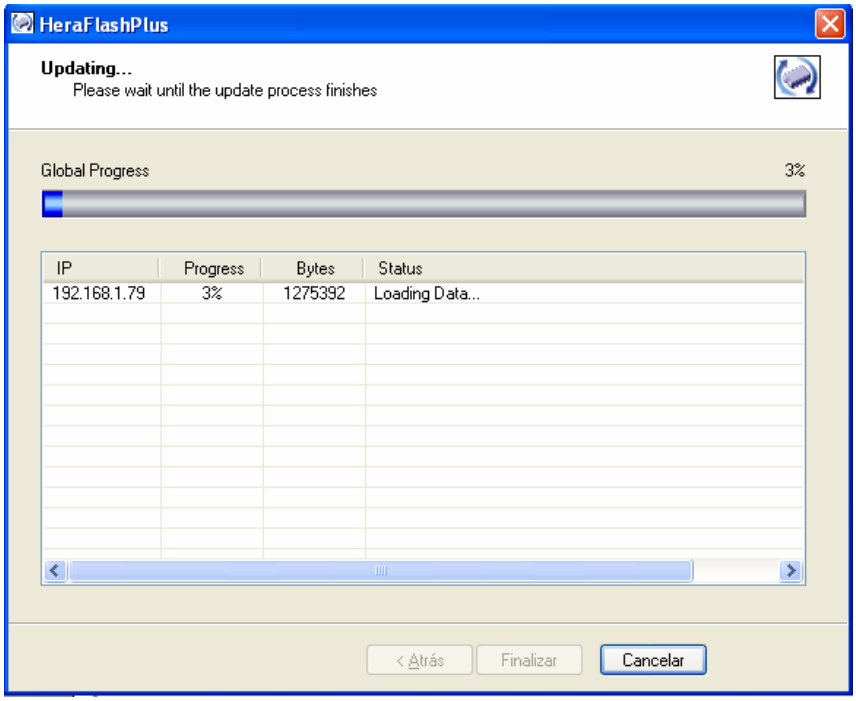
3. All the units connected to the computer's network will be detected automatically. Select which units will be upgraded.



- Press next to start the update process. You will see a warning message telling you that all the contents of the Flash will be erased and that you must back-up this data BEFORE updating the device.

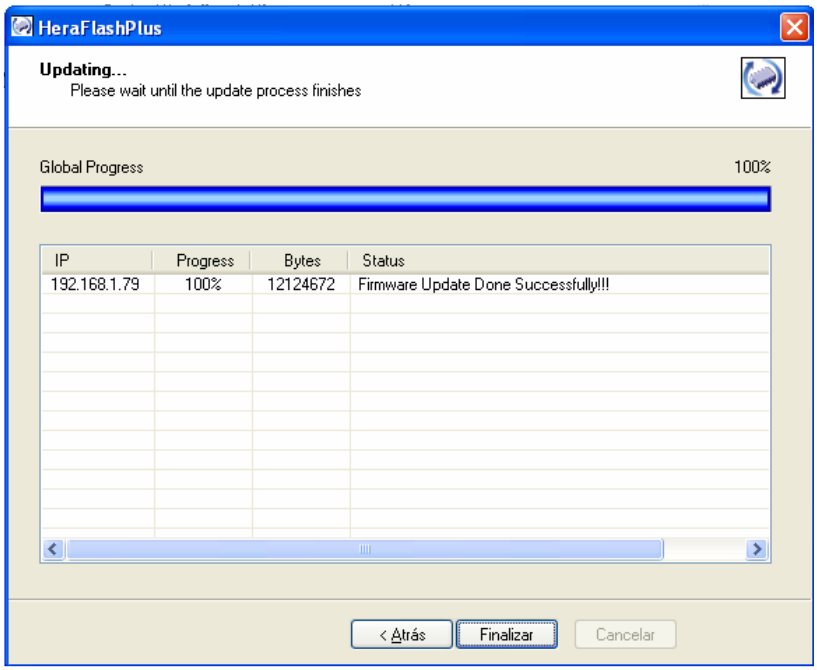
HeraFlashPlus2.5.0
WARNING!!!! All the FLASH contents will be erased. Please make a backup if needed before continuing.
Do you want to continue updating erasing FLASH contents?
Aceptar
Cancelar

- By clicking on next, the upgrading process will start.



The update process will begin, the Flash will be erased and a progress bar will tell you more info and alert you when it is finished.

- You are able to abort the update process until the recording process begins. At this point, the cancel button is disabled. In any case, even if a power failure or a similar failure takes place, it is always possible to return to start the recording process again.
- If everything has gone correctly, at the end of the recording, you will be shown the following message:



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